



STIC Search Report

EIC 2600

STIC Database Tracking Number: 109437

TO: Dmitry Brant
Location: Pk2 3C16
Art Unit: 2655
Wednesday, December 03, 2003
Case Serial Number: 09/826715

From: Pamela Reynolds
Location: EIC 2600
PK2-3C03
Phone: 306-0255

Pamela.Reynolds@uspto.gov

Search Notes

Dear Dmitry Brant,

Please find attached the search results for 09/826715. I used the search strategy I emailed to you to edit, not hearing from you I proceeded. I searched the standard Dialog files, IBM TDBs, and IEEE.

If you would like a re-focus please let me know.

Thank you.

Pamela Reynolds

File 344:Chinese Patents Abs Aug 1985-2003/Apr
(c) 2003 European Patent Office
File 347:JAPIO Oct 1976-2003/Jul(Updated 031105)
(c) 2003 JPO & JAPIO
File 348:EUROPEAN PATENTS 1978-2003/Nov W04
(c) 2003 European Patent Office
File 349:PCT FULLTEXT 1979-2002/UB=20031127,UT=20031120
(c) 2003 WIPO/Univentio
File 350:Derwent WPIX 1963-2003/UD,UM &UP=200376
(c) 2003 Thomson Derwent

Set	Items	Description
S1	268	AU=(SHU, C? OR SHU, H? OR SHU C? OR SHU H?)
S2	1	S1 AND SPEECH()RECOG?

/5,K/1 (Item 1 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2003 Thomson Derwent. All rts. reserv.

012964689 **Image available**
WPI Acc No: 2000-136540/200012
XRPX Acc No: N00-102113

Rejection grammar processing method in utterance speech recognition system
Patent Assignee: GTE INTERNETWORKING INC (SYLV)
Inventor: SHU C
Number of Countries: 001 Number of Patents: 001
Patent Family:
Patent No Kind Date Applcat No Kind Date Week
US 6016470 A 20000118 US 97969031 A 19971112 200012 B

Priority Applications (No Type Date): US 97969031 A 19971112

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
US 6016470	A	14		G10L-005/06	

Abstract (Basic): US 6016470 A

NOVELTY - A digitized sub-list of phoneme models is selected from a list of phoneme models and a digitized sequential representation is presented. A set of probabilities indicating how well the sequential representation matches sequential combination of sublist is calculated.

DETAILED DESCRIPTION - The digitized sequential representation of utterance is compared with sequence of generated digitized list of words using an accepted main grammar process and a second set of probabilities is calculated. The highest probabilities for each of the two sets is determined. If the highest probability is found in first set of probabilities, the utterance is rejected and if found in the second set, the utterance is accepted. The selected digitized list is generated by compiling a complete list of all phoneme models in a language and forming a test set of digitized sequential representation of utterance with acceptable and rejectable parts. The results of test set are analyzed and a false rejection list, false acceptance list and their statistics are accumulated. If the statistics of the process is acceptable, the list of models are used and if not acceptable, the models are further processed to become acceptable. INDEPENDENT CLAIMS are also included for the following:

- (a) Computer system for speech recognition ;
- (b) speech recognition program product

USE - In utterance speech recognition systems.

ADVANTAGE - The main grammars are with large vocabularies of about thirty phonemes. The system requires only small memories thereby reducing cost and speech recognition is accurate and faster.

DESCRIPTION OF DRAWING(S) - The figure shows the flowchart of method of selecting phonemes.

pp; 14 DwgNo 5/7

Title Terms: REJECT; GRAMMAR; PROCESS; METHOD; SPEECH; RECOGNISE; SYSTEM

Derwent Class: P86; T01; W04

International Patent Class (Main): G10L-005/06

File Segment: EPI; EngPI

Rejection grammar processing method in utterance speech recognition system

Inventor: SHU C

Abstract (Basic):

... a) Computer system for **speech recognition** ;
(...)

...b) **speech recognition** program product...

...In utterance **speech recognition** systems...

...vocabularies of about thirty phonemes. The system requires only small memories thereby reducing cost and **speech recognition** is accurate and faster

?

File 344:Chinese Patents Abs Aug 1985-2003/Apr
(c) 2003 European Patent Office
File 347:JAPIO Oct 1976-2003/Jul(Updated 031105)
(c) 2003 JPO & JAPIO
File 350:Derwent WPIX 1963-2003/UD,UM &UP=200377
(c) 2003 Thomson Derwent

? ds

Set	Items	Description
S1	36	CEPSTRAL(3N)COEFFICIENT?
S2	18	(COMPAR? OR EVALUAT? OR DETERMIN? OR ASSESS? OR DISCERN? OR DISTINGUISH?) AND S1
S3	108998	PEAK?
S4	13002	(MANY OR MULTIPLE OR SEVERAL OR MULTI OR NUMEROUS OR PLURAL?) AND S3
S5	898425	FRAME??
S6	36487	SPEECH(3N)RECOG?
S7	506	ACOUSTIC(3N)DATA AND (PARTITION? OR DIVID? OR SEPARAT? OR - DIVISION? OR PART OR PARTS OR SECTION?? OR SEGMENT?? OR PORTION?? OR FRAGMENT? OR PIECES OR SECTOR??)
S8	49412	IC=G10L?
S9	0	S2 AND S4
S10	7	S2 AND S5
S11	3689	(COMPAR? OR EVALUAT? OR DETERMIN? OR ASSESS? OR DISCERN? OR DISTINGUISH?) AND S4
S12	605	FIRST AND S11 AND SECOND
S13	26	S12 AND S5
S14	1	S13 AND S7
S15	1	S14 NOT S10
S16	220	FIRST AND SECOND AND S5 AND S3
S17	0	S16 AND S1
S18	1	S16 AND S7
S19	0	S18 NOT (S14 OR S10)
S20	6	S16 AND S8
S21	5	S20 NOT (S14 OR S10)
S22	3	S7 AND S4
S23	2	S22 NOT (S20 OR S14 OR S10)

10/3,K/1 (Item 1 from file: 350)

DIALOG(R)File 350:Derwent WPIX

(c) 2003 Thomson Derwent. All rts. reserv.

014045194 **Image available**

WPI Acc No: 2001-529407/200158

XRPX Acc No: N01-392952

Speech signal encoding technique uses analysis of fundamental frequency and harmonics to provide quality reproduction

Patent Assignee: MATRA NORTEL COMMUNICATIONS (NELE); MATRA NORTEL COMMUNICATIONS SAS (NELE)

Inventor: CAPMAN F; MURGIA C

Number of Countries: 095 Number of Patents: 004

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 200103120	A1	20010111	WO 2000FR1908	A	20000704	200158 B
AU 200062922	A	20010122	AU 200062922	A	20000704	200158
FR 2796190	A1	20010112	FR 998634	A	19990705	200158
EP 1192621	A1	20020403	EP 2000949623	A	20000704	200230
			WO 2000FR1908	A	20000704	

Priority Applications (No Type Date): FR 998634 A 19990705

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

WO 200103120 A1 F 58 G10L-019/02

Designated States (National): AE AG AL AM AT AU AZ BA BB BG BR BY BZ CA CH CN CR CU CZ DE DK DM DZ EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR LS LT LU LV MA MD MG MK MN MW MX MZ NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT TZ UA UG US UZ VN YU ZA ZW

Designated States (Regional): AT BE CH CY DE DK EA ES FI FR GB GH GM GR IE IT KE LS LU MC MW MZ NL OA PT SD SE SL SZ TZ UG ZW

AU 200062922 A G10L-019/02 Based on patent WO 200103120

FR 2796190 A1 G10L-019/02

EP 1192621 A1 F G10L-019/02 Based on patent WO 200103120

Designated States (Regional): AL AT BE CH CY DE DK ES FI FR GB GR IE IT LI LT LU LV MC MK NL RO SI

Abstract (Basic):

... audio encoding technique, the encoder estimates the fundamental frequency (F0) of an audio signal, and determines a spectrum of the audio signal by a transform of a frame of the audio signal in the frequency domain. Data representing the spectral amplitudes associated with...

...module in the neighborhood of this frequency multiple. Data is obtained by use of the cepstral coefficients, calculated by transforming in the cepstral domain a compressed higher envelope of the spectrum of...

10/3,K/2 (Item 2 from file: 350)

DIALOG(R)File 350:Derwent WPIX

(c) 2003 Thomson Derwent. All rts. reserv.

013966916 **Image available**

WPI Acc No: 2001-451130/200148

XRPX Acc No: N01-334027

Audio encoding process for speech transmission includes use of cepstral coefficients and interpolation in decoding

Patent Assignee: MATRA NORTEL COMMUNICATIONS (NELE); MATRA NORTEL COMMUNICATIONS SAS (NELE)

Inventor: CAPMAN F; MURGIA C
Number of Countries: 095 Number of Patents: 004

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 200103118	A1	20010111	WO 2000FR1906	A	20000704	200148 B
AU 200062920	A	20010122	AU 200062920	A	20000704	200148
FR 2796191	A1	20010112	FR 998635	A	19990705	200148
EP 1192619	A1	20020403	EP 2000949621	A	20000704	200230
			WO 2000FR1906	A	20000704	

Priority Applications (No Type Date): FR 998635 A 19990705

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
WO 200103118	A1	F	52	G10L-019/02	
Designated States (National): AE AG AL AM AT AU AZ BA BB BG BR BY BZ CA CH CN CR CU CZ DE DK DM DZ EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR LS LT LU LV MA MD MG MK MN MW MX MZ NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT TZ UA UG US UZ VN YU ZA ZW					
Designated States (Regional): AT BE CH CY DE DK EA ES FI FR GB GH GM GR IE IT KE LS LU MC MW MZ NL OA PT SD SE SL SZ TZ UG ZW					
AU 200062920	A			G10L-019/02	Based on patent WO 200103118
FR 2796191	A1			G10L-019/02	
EP 1192619	A1	F		G10L-019/02	Based on patent WO 200103118
Designated States (Regional): AL AT BE CH CY DE DK ES FI FR GB GR IE IT LI LT LU LV MC MK NL RO SI					

Audio encoding process for speech transmission includes use of cepstral coefficients and interpolation in decoding

Abstract (Basic):

... The method includes use of a decoder to synthesise a set of successive frames of N sample of an audio signal from encoding data which is includes a in a digital flow received from the encoder. For only one subset of frames this includes data representing spectral amplitudes of the audio signal. For each of the frames of the subset the decoder determines the cepstral coefficients ($cxq(n)$) representing at least some of the spectral amplitudes. For frames not forming part of the subset, it interpolates the cepstral coefficients and generates a spectral estimate of the audio signal which it transforms in the temporal domain to obtain the synthesised frame.

10/3,K/3 (Item 3 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2003 Thomson Derwent. All rts. reserv.

011526848 **Image available**
WPI Acc No: 1997-503334/199746
XRPX Acc No: N97-419499

Feature extractor for automated speech system - calculate logarithm of input frame spectrum and cepstrum of this logarithm, also detecting cepstral coefficient meeting predetermined criterion, and derives feature of detected cepstral coefficient representing voiced speech frame

Patent Assignee: BRITISH TELECOM PLC (BRTE)

Inventor: POWER K J; RINGLAND S P A

Number of Countries: 027 Number of Patents: 002

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 9737345	A1	19971009	WO 97GB816	A	19970324	199746 B
AU 9721669	A	19971022	AU 9721669	A	19970324	199808

Priority Applications (No Type Date): EP 96302235 A 19960329

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
-----------	------	-----	----	----------	--------------

WO 9737345	A1	E	24	G10L-009/00	
------------	----	---	----	-------------	--

Designated States (National): AU CA CN JP KR MX NO NZ SG US

Designated States (Regional): AT BE CH DE DK ES FI FR GB GR IE IT LU MC
NL PT SE

AU 9721669	A	G10L-009/00	Based on patent WO 9737345
------------	---	-------------	----------------------------

... calculate logarithm of input frame spectrum and cepstrum of this logarithm, also detecting cepstral coefficient meeting predetermined criterion, and derives feature of detected cepstral coefficient representing voiced speech frame

...Abstract (Basic): The feature extractor receives an input digital signal which is divided into frames, and calculates the logarithm of the spectrum of an input frame. A cepstrum calculator (334) calculates the cepstrum of the logarithm of the spectrum of the frame.

...

...A pitch detector (335) detects a cepstral coefficient meeting a predetermined criterion. A feature deriver (336) derives a feature relating to the detected cepstral coefficient, which represents whether the input frame includes voiced speech. The spectrum calculator evaluates the logarithm of the power spectrum of the input signal...

...speech processing. For extracting features from input signal for use by subsequent automated speech systems. Cepstral coefficients lying inside normal speech frequency range, including first twenty cepstral coefficients, may be discarded

...Title Terms: FRAME ;

10/3,K/4 (Item 4 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2003 Thomson Derwent. All rts. reserv.

010410317 **Image available**

WPI Acc No: 1995-311666/199540

XRPX Acc No: N95-235353

Adaptive method for speaker identification and verification - converting audio input to frames that have adaptive weighing component applied for normalisation prior to recognition processing

Patent Assignee: UNIV RUTGERS STATE NEW JERSEY (RUTF)

Inventor: ASSALEH K T; MAMMONE R J

Number of Countries: 060 Number of Patents: 009

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 9523408	A1	19950831	WO 95US2801	A	19950228	199540 B
AU 9521164	A	19950911	AU 9521164	A	19950228	199550
US 5522012	A	19960528	US 94203988	A	19940228	199627
EP 748500	A1	19961218	EP 95913980	A	19950228	199704
			WO 95US2801	A	19950228	
AU 683370	B	19971106	AU 9521164	A	19950228	199802
JP 10500781	W	19980120	JP 95522534	A	19950228	199813
			WO 95US2801	A	19950228	

MX 9603686	A1	19971201	MX 963686	A	19960827	199936
CN 1142274	A	19970205	CN 95191853	A	19950228	200053
MX 194244	B	19991124	MX 963686	A	19950228	200106

Priority Applications (No Type Date): US 94203988 A 19940228

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
-----------	------	-----	----	----------	--------------

WO 9523408	A1	E	34	G10L-005/06	
------------	----	---	----	-------------	--

Designated States (National): AM AT AU BB BG BR BY CA CH CN CZ DE DK ES FI GB GE HU JP KE KG KP KR KZ LK LT LU LV MD MG MN MW MX NL NO NZ PL PT RO RU SD SE SG SI SK TJ TT UA UG UZ VN

Designated States (Regional): AT BE CH DE DK ES FR GB GR IE IT LU MC NL OA PT SE

AU 9521164	A		G10L-005/06	Based on patent WO 9523408
------------	---	--	-------------	----------------------------

US 5522012	A	13	G10L-005/06	
------------	---	----	-------------	--

EP 748500	A1	E	34	G10L-005/06	Based on patent WO 9523408
-----------	----	---	----	-------------	----------------------------

Designated States (Regional): AT BE CH DE DK ES FR GB GR IE IT LI LU MC NL PT SE

AU 683370	B		G10L-005/06	Previous Publ. patent AU 9521164
-----------	---	--	-------------	----------------------------------

Based on patent WO 9523408

JP 10500781	W	36	G10L-003/00	Based on patent WO 9523408
-------------	---	----	-------------	----------------------------

MX 9603686	A1		G10L-005/06	
------------	----	--	-------------	--

CN 1142274	A		G10L-005/06	
------------	---	--	-------------	--

MX 194244	B		G10L-005/006	
-----------	---	--	--------------	--

... converting audio input to frames that have adaptive weighing component applied for normalisation prior to recognition processing

...Abstract (Basic): over a channel such as a telephone line. The input is digitised and converted to frames that are analysed by linear prediction. This extracts prediction coefficients...

...Prediction coefficients are derived from the normalised speech to allow for identification. The pattern is compared to a number of speech patterns produced by a number of persons in advance...

...Abstract (Equivalent): windowing a speech segment into a plurality of speech frames ;

...

... determining linear prediction coefficients from a linear predictive polynomial for each said frame of speech...

... determining a first cepstral coefficient from said linear prediction coefficients in which first cepstrum information comprises said first cepstral coefficient ;

...

... determining a plurality of roots of said linear prediction polynomial from the poles of said all...

...selecting one of said frames having a predetermined number of said roots within a unit circle of the z-plane in which said selected frames form said predetermined components of said first cepstrum information...

...an adaptive component weighting cepstrum to attenuate broad bandwidth components in said speech signal, by determining a finite impulse response filter for emphasizing the speech formants of said speech signal and attenuating said residue components comprising the steps of determining a finite impulse response filter for emphasizing the speech formants of said speech signal and attenuating said residue

components, determining adaptive component weighting coefficients from said finite impulse response filter, determining a second cepstral coefficient from said adaptive component weighting coefficients, and subtracting said second cepstral coefficient from said first cepstral coefficient for forming said adaptive component weighting cepstrum; and
...Title Terms: **FRAME** ;

10/3,K/5 (Item 5 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2003 Thomson Derwent. All rts. reserv.

009360797 **Image available**
WPI Acc No: 1993-054275/199307
XRPX Acc No: N93-041415
Mobile telephone speech transmission appts. with speech recognition control - uses signal processing circuit to convert GSM coding parameters into speech recognition parameters

Patent Assignee: PHILIPS PATENTVERWALTUNG GMBH (PHIG); PHILIPS GLOEILAMPENFAB NV (PHIG)

Inventor: HIRSCH H; RUEHL H

Number of Countries: 004 Number of Patents: 004

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
EP 527535	A2	19930217	EP 92202424	A	19920806	199307 B
DE 4126882	A1	19930218	DE 4126882	A	19910814	199308
JP 5241590	A	19930921	JP 92215975	A	19920813	199342
EP 527535	A3	19931020	EP 92202424	A	19920806	199510

Priority Applications (No Type Date): DE 4126882 A 19910814

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
EP 527535	A2	G	12	G10L-005/06	
Designated States (Regional): DE FR GB					
DE 4126882	A1	G10L-005/06			
JP 5241590	A	G10L-003/00			
EP 527535	A3	G10L-005/06			

...Abstract (Basic): mobile telephone includes a speech coder (3) for encoding digital speech signals within a time **frame** using coding parameters. A signal processing circuit (6) receives part of the set of coding...

...The evaluating circuit computes logarithmic area ratio coefficients into Cepstral coefficients from the coding parameters. The evaluating circuit determines speech parameters by comparing sets of coding parameters containing long term prediction gain factors with a selected threshold value...

10/3,K/6 (Item 6 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2003 Thomson Derwent. All rts. reserv.

008857962 **Image available**
WPI Acc No: 1991-361985/199150
XRPX Acc No: N91-277335
Rejection method for speech recognition - deriving parameters derived from sequence frames with best choice matches to

references used for unknow

Patent Assignee: LENNIG M (LENN-I); NORTHERN TELECOM LTD (NELE)

Inventor: LENNIG M

Number of Countries: 002 Number of Patents: 003

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
CA 2013263	A	19910928	CA 2013263	A	19900328	199150 B
US 5097509	A	19920317	US 90501993	A	19900328	199214
CA 2013263	C	19950905	CA 2013263	A	19900328	199542

Priority Applications (No Type Date): CA 2013263 A 19900328; US 90501993 A 19900328

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
US 5097509	A		11		
CA 2013263	C			G10L-005/06	

... deriving parameters derived from sequence frames with best choice matches to references used for unknow

...Abstract (Basic): method for speech recognition comprises representing an unknown utterance as a first sequence of parameter **frames** . Each parameter **frame** includes a set of primary and secondary parameters and an equalised second sequence of parameter **frames** derived from the first sequence of parameters **frames** . Each of the primary and secondary parameters in the sequence of parameter **frames** of the representation of the unknown utterance are **compared** to each of a number of reference representations expressed in the same kind of parameters, to **determine** how closely each reference representation resembles the representation of the unknown utterance...

...Abstract (Equivalent): differences between pairs of primary cepstra. The equalised representation being the signed difference of each **cepstral coefficient** less an average value of the coefficients...

...Factors are generated from the ordered lists of templates to **determine** the probability of the top choice being a correct acceptance, with different methods, being a...

...Title Terms: **FRAME** ;

10/3,K/7 (Item 7 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2003 Thomson Derwent. All rts. reserv.

007775698

WPI Acc No: 1989-040810/198906

XRPX Acc No: N89-031227

Speech recognition system using linear predictive coding - compares received speech frames with reference templates, generates error valves, and selects words with small error valves

Patent Assignee: TEXAS INSTR INC (TEXI)

Inventor: ANDERSON W; DODDINGTON G R; MCMAHAN M L; RAJASEKARAN P K; RAJASEKARA P K

Number of Countries: 006 Number of Patents: 005

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
EP 302663	A	19890208	EP 88306967	A	19880728	198906 B
US 4910784	A	19900320	US 8779563	A	19870730	199017
EP 302663	B1	19931013	EP 88306967	A	19880728	199341
DE 3884880	G	19931118	DE 3884880	A	19880728	199347
			EP 88306967	A	19880728	

KR 123934

B1 19971126 KR 889623

A 19880729 199950

Priority Applications (No Type Date): US 8779563 A 19870730

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

EP 302663 A E 9

Designated States (Regional): DE FR GB IT

US 4910784 A 9

EP 302663 B1 E 13 G10L-005/06

Designated States (Regional): DE FR GB IT

DE 3884880 G G10L-005/06 Based on patent EP 302663

KR 123934 B1 G10L-005/06

... compares received speech frames with reference templates, generates error valves, and selects words with small error valves

...Abstract (Basic): a digital representation. A feature extractor coupled to the digitizer groups the digital signals into **frames** and generates a transform of the signal of each **frame**. The transform has a number of feature coefficients, and each feature coefficient has a corresponding...

...than a preselected threshold for that coefficient. A queue coupled to the feature extractor receives **frames** of binary feature coefficients and arranges them in consecutive order...

...A **comparator** coupled to the queue **compares** several speech **frames** with a number of reference templates having **frames** of binary feature coefficients and generates a number of error values indicating the closeness of the match between them. A decision controller coupled to the **comparator** receives the results of the **comparisons**, and selects a best match between a portion of a speech utterance and the reference

...Abstract (Equivalent): 18) coupled to said A/D converter (16) for grouping the speech samples into speech **frames** and generating LPC parameters for each said speech **frame**, transforming said LPC parameters into cepstral parameters and deriving a **frame** of binary feature **coefficients** by coding said **cepstral** parameters into binary values each indicating a value greater or less than a preselected threshold and grouping the said binary values into said **frames** of binary features coefficients, a push-down queue (40) coupled to said feature extractor for receiving successive **frames** of binary feature coefficients corresponding to the successive speech **frames**; a **comparator** (20) coupled to said queue (40) for **comparing** a plurality of the last received said **frames** of binary feature coefficients with a plurality of reference templates (22) each consisting of a plurality of **frames** of binary coefficients and generating a plurality of error values indicating the closeness of the match therebetween wherein only alternate **frames** in said queue (40) are used by said **comparator** (20) for the **comparison** with the templates (22), the number of said alternate **frames** being a function of the template length, and a decision controller (24) coupled to said **comparator** for receiving the results of the **comparisons**, and for selecting a best match between a portion of the speech signal and the...

...Abstract (Equivalent): speech signal. A feature extractor coupled to the digitiser groups the digital speech signals into **frames** and generates a transform of the digital speech signals as grouped in each **frame**. The transform has a number of feature coefficients, and each feature coefft. has a corresp...

...A queue is coupled to the feature extractor to receive **frames** of

binary feature coefficients as speech **frames** and arranged them in consecutive order. A **comparator** is coupled to the queue to **compare** a number of speech **frames** with a number of reference templates that have **frames** of binary feature coefficients and generates a number of error values indicating the closeness of...

...The reference templates are representative of different words. A decision controller is coupled to the **comparator** to receive the results of the **comparisons**, and to select a best match between a portion of a speech utterance as represented by the speech **frames** and the reference templates...

...USE/ADVANTAGE - With main **frame**, mini- or micro computer. Flexible and accurate vocabulary enrolment.

...Title Terms: **COMPARE** ;

?

15/3,K/1 (Item 1 from file: 350)

DIALOG(R)File 350:Derwent WPIX

(c) 2003 Thomson Derwent. All rts. reserv.

010012599 **Image available**

WPI Acc No: 1994-280310/199435

Related WPI Acc No: 1995-157069; 1995-225836; 1995-294609; 1995-383239;
1996-097792; 1996-260111; 1996-260116; 1996-278120; 1997-119282

XRXPX Acc No: N94-220925

Multifunction communication system for use with personal computer -
includes packet protocol for communications between software components
running on personal computer and local hardware components over serial
communications link

Patent Assignee: SHARMA R (SHAR-I); MULTI-TECH SYSTEMS INC (MULT-N)

Inventor: DAVIS J P; GUNN T D; LI P; MAITRA S; SHARMA R; THANAWALA A; YOUNG
S

Number of Countries: 020 Number of Patents: 017

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
CA 2104701	A	19940709	CA 2104701	A	19930824	199435	B
EP 630141	A2	19941221	EP 93403164	A	19931223	199504	
US 5452289	A	19950919	US 932467	A	19930108	199543	
US 5471470	A	19951128	US 932467	A	19930108	199602	
			US 94289294	A	19940811		
US 5500859	A	19960319	US 932467	A	19930108	199617	
			US 94289304	A	19940811		
EP 630141	A3	19960703	EP 93403164	A	19931223	199636	
US 5559793	A	19960924	US 932467	A	19930108	199644	
			US 94289305	A	19940811		
US 5574725	A	19961112	US 932467	A	19930108	199651	
			US 94289295	A	19940811		
US 5577041	A	19961119	US 932467	A	19930108	199701	
			US 94289294	A	19940811		
			US 95488183	A	19950607		
US 5592586	A	19970107	US 932467	A	19930108	199708	
			US 94289297	A	19940811		
US 5600649	A	19970204	US 932467	A	19930108	199711	
			US 95527849	A	19950914		
US 5673257	A	19970930	US 932467	A	19930108	199745	
			US 95428904	A	19950425		
US 5673268	A	19970930	US 932467	A	19930108	199745	
			US 94289296	A	19940811		
JP 9238200	A	19970909	JP 93251131	A	19930913	199746	
US 5764627	A	19980609	US 932467	A	19930108	199830	
			US 95488183	A	19950607		
			US 96636582	A	19960423		
US 5790532	A	19980804	US 932467	A	19930108	199838	
			US 95527952	A	19950914		
CA 2104701	C	20021112	CA 2104701	A	19930824	200302	

Priority Applications (No Type Date): US 932467 A 19930108; US 94289294 A
19940811; US 94289304 A 19940811; US 94289305 A 19940811; US 94289295 A
19940811; US 95488183 A 19950607; US 94289297 A 19940811; US 95527849 A
19950914; US 95428904 A 19950425; US 94289296 A 19940811; US 96636582 A
19960423; US 95527952 A 19950914

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

CA 2104701 A 161 H04L-005/22

EP 630141 A2 E 99 H04M-003/42

Designated States (Regional): AT BE CH DE DK ES FR GB GR IE IT LI LU MC
NL PT SE

US 5452289	A	79 H04B-003/23	
US 5471470	A	79 H04J-003/17	Div ex application US 932467
US 5500859	A	81 H04J-003/17	Div ex application US 932467
			Div ex patent US 5452289
EP 630141	A3	H04L-005/22	
US 5559793	A	80 H04B-003/23	Div ex application US 932467
			Div ex patent US 5452289
US 5574725	A	80 H04J-003/12	Div ex application US 932467
			Div ex patent US 5452289
US 5577041	A	80 H04M-011/00	Cont of application US 932467
			Div ex application US 94289294
			Cont of patent US 5452289
			Div ex patent US 5471470
US 5592586	A	79 G10L-009/00	Div ex application US 932467
			Div ex patent US 5452289
US 5600649	A	80 H04J-003/17	Div ex application US 932467
			Div ex patent US 5452289
US 5673257	A	79 H04B-003/23	Div ex application US 932467
			Div ex patent US 5452289
US 5673268	A	79 H04J-003/12	Div ex application US 932467
			Div ex patent US 5452289
JP 9238200	A	61 H04M-011/00	Cont of application US 932467
US 5764627	A	H04M-001/00	Cont of application US 95488183
			Cont of patent US 5452289
			Cont of patent US 5577041
US 5790532	A	H04J-003/16	Div ex application US 932467
			Div ex patent US 5452289

CA 2104701 C E H04L-005/22
...Abstract (Equivalent): calling a remote modem from a local modem over a telephone line connection, line a **portion** of which includes a cellular telephone link connection compression **section** having means for...

...e.) compressing the outgoing digital voice data into compressed outgoing digital voice data **frames**,

...

...h.) decompressing the compressed incoming digital voice data **frames** into the incoming digital voice data...

...2.) the memory connected to the voice compression **section** and to a data transmission **section** ; and...

...3.) the data transmission **section** having means for...

...a.) receiving the compressed outgoing digital voice data **frames** from the memory...

...c.) placing the compressed outgoing digital voice data **frames** into compressed outgoing digital voice data packets... **determining** a power of a at least a **portion** of the compressed outgoing digital voice data packet including a **plurality** of samples of the local voice signals as a function of the summation of the square of each sample over the **portion** of the compressed outgoing digital voice data packet; and...

... **comparing** the power of the **portion** of the compressed outgoing digital voice data packets to a preselected threshold to indicate whether...voice signals into discrete samples of digital voice data and collecting the discrete samples into **segments** ;

...means for **dividing** the **segments** into subsegments and for producing therefrom a current voice subsegment...

...pitch prediction means for **determining** the long term predicted gain of the current voice subsegment by **comparing** the current voice subsegment to reconstructed voice samples to produce a pitch predictor gain and...means for **determining** the **peak** amplitude of the long term residual samples...

...means for scaling the long term residual samples based on the **peak** amplitude to produce normalized long term residual samples...

...means including a code book stored in a memory for **comparing** the normalized long term residual samples to stored distinct normalized long term residual samples stored...

...for providing the distinct memory address, the pitch predictor gain, the lag component and the **peak** amplitude for each voice subsegment... creating a qualified packet having a qualified packet identifier and a **plurality** of command identifiers for communicating control information ...if in voice over data mode of operation, the acoustic echo cancellation means including a **first** Finite Impulse Response filter and operable in response to the selected line echo cancelled incoming digital voice data or the selected decompressed incoming digital voice data for removing **acoustic** echo from the outgoing digital voice data and for producing therefrom acoustic echo cancelled outgoing data, the line echo cancellation means including a **second** Finite Impulse Response filter and operable in response to the acoustic echo cancelled outgoing digital...

?

21/3,K/1 (Item 1 from file: 350)

DIALOG(R)File 350:Derwent WPIX

(c) 2003 Thomson Derwent. All rts. reserv.

011457465 **Image available**

WPI Acc No: 1997-435372/199740

XRPX Acc No: N97-362107

Speech waveform pitch estimation method - using vocoder to compare one or more correlation peaks with clipping threshold value, so that additional calculations may be performed if single peak is greater than clipping threshold

Patent Assignee: ADVANCED MICRO DEVICES INC (ADMI)

Inventor: BARTKOWIAK J G

Number of Countries: 019 Number of Patents: 005

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 9731366	A1	19970828	WO 97US1281	A	19970124	199740 B
EP 882287	A1	19981209	EP 97904886	A	19970124	199902
			WO 97US1281	A	19970124	
US 5864795	A	19990126	US 96603366	A	19960220	199911
EP 882287	B1	20010912	EP 97904886	A	19970124	200155
			WO 97US1281	A	19970124	
DE 69706650	E	20011018	DE 606650	A	19970124	200169
			EP 97904886	A	19970124	
			WO 97US1281	A	19970124	

Priority Applications (No Type Date): US 96603366 A 19960220

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

WO 9731366 A1 E 31 G10L-003/00

Designated States (National): JP

Designated States (Regional): AT BE CH DE DK ES FI FR GB GR IE IT LU MC
NL PT SE

EP 882287 A1 E G10L-003/00 Based on patent WO 9731366

Designated States (Regional): BE DE GB

US 5864795 A G10L-009/08

EP 882287 B1 E G10L-011/04 Based on patent WO 9731366

Designated States (Regional): BE DE GB

DE 69706650 E G10L-011/04 Based on patent EP 882287

Based on patent WO 9731366

... using vocoder to compare one or more correlation peaks with clipping threshold value, so that additional calculations may be performed if single peak is greater than clipping threshold

...Abstract (Basic): The method involves performing a correlation calculation on a first frame of a speech waveform. The correlation calculation for the first frame produces one or more correlation peaks at respective numbers of the delay samples. A single correlation peak is determined from the one or more correlation peaks. The single peak has a peak location (Pd) comprising a first number of delay samples...

...The method further involves searching for a peak location (Pd') where the single peak location Pd of the signal correlation peak is a multiple of the peak location Pd'. The peak location Pd' has a correlation peak. The peak location Pd' comprises a second number of delay samples. Finally the pitch is set equal to the second number of delay samples indicated by the peak location Pd...

...ADVANTAGE - More accurate estimate of pitch of received waveform.

Disregards second and higher multiples of true pitch, more accurately
...
...Title Terms: **PEAK** ;
International Patent Class (Main): **G10L-003/00** ...
... **G10L-009/08** ...
... **G10L-011/04**

21/3,K/2 (Item 2 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2003 Thomson Derwent. All rts. reserv.

010463055 **Image available**
WPI Acc No: 1995-364374/199547
Processing speech signal for voice synthesis system - deepening concave
part between spectral formants and emphasising peaks
Patent Assignee: SONY CORP (SONY)
Inventor: MATSUMOTO J; NISHIGUCHI M
Number of Countries: 002 Number of Patents: 003
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 7248794	A	19950926	JP 9439979	A	19940310	199547 B
US 5953696	A	19990914	US 95398363	A	19950303	199944
			US 97935695	A	19970923	
JP 3321971	B2	20020909	JP 9439979	A	19940310	200264

Priority Applications (No Type Date): JP 9439979 A 19940310

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
JP 7248794	A	13		G10L-007/02	
US 5953696	A			G10L-009/02	Cont of application US 95398363
JP 3321971	B2	13		G10L-013/00	Previous Publ. patent JP 7248794

... deepening concave part between spectral formants and emphasising peaks

...Abstract (Basic): The speech signal processing method consists of several steps. First, the signal (S1) is subjected to a high pass formant emphasis process. Next, a second emphasis process is applied to the speech signal to extend the entire region, more specifically...

...the valley of the frequency. Subsequently, a third emphasis process is applied to emphasize the peak magnitude of a formant in the voice frame in the upright part of the speech signal (S3). Finally, a fourth emphasis process is...

...Title Terms: **PEAK**

International Patent Class (Main): **G10L-007/02** ...

... **G10L-009/02** ...

... **G10L-013/00**

21/3,K/3 (Item 3 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2003 Thomson Derwent. All rts. reserv.

010420897 **Image available**
WPI Acc No: 1995-322213/199542

XRPX Acc No: N95-242503

Excitation signal synthesis for frame erasure of packet data loss e.g. for speech coding system - synthesising linear prediction filter coeffs. during erased frames using weighting extrapolation to obtain bandwidth expansion of peaks in filter response

Patent Assignee: AT & T CORP (AMTT); AMERICAN TELEPHONE & TELEGRAPH CO (AMTT); LUCENT TECHNOLOGIES INC (LUCE)

Inventor: CHEN J

Number of Countries: 009 Number of Patents: 010

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
EP 673017	A2	19950920	EP 95301298	A	19950228	199542 B
AU 9513673	A	19950921	AU 9513673	A	19950307	199545
JP 7311597	A	19951128	JP 9579358	A	19950313	199605
CA 2142393	A	19950915	CA 2142393	A	19950213	199606
US 5615298	A	19970325	US 94212408	A	19940314	199718
EP 673017	A3	19970813	EP 95301298	A	19950228	199745
CA 2142393	C	19990119	CA 2142393	A	19950213	199914
JP 3439869	B2	20030825	JP 9579358	A	19950313	200357
EP 673017	B1	20030903	EP 95301298	A	19950228	200360
DE 69531642	E	20031009	DE 631642	A	19950228	200374
			EP 95301298	A	19950228	

Priority Applications (No Type Date): US 94212408 A 19940314

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
EP 673017	A2	E	97	G10L-009/14	
	Designated States (Regional):	DE	ES	FR	GB IT
AU 9513673	A			G10L-005/02	
JP 7311597	A		14	G10L-009/14	
CA 2142393	A			G10L-009/00	
US 5615298	A		60	G10L-003/02	
EP 673017	A3			G10L-009/14	
CA 2142393	C			G10L-009/00	
JP 3439869	B2		14	G10L-019/04	Previous Publ. patent JP 7311597
EP 673017	B1	E		G10L-019/00	
	Designated States (Regional):	DE	ES	FR	GB IT
DE 69531642	E			G10L-019/00	Based on patent EP 673017
	Excitation signal synthesis for frame erasure of packet data loss e.g. for speech coding system...				
	...synthesising linear prediction filter coeffs. during erased frames using weighting extrapolation to obtain bandwidth expansion of peaks in filter response				

...Abstract (Basic): method for a signal imitating human speech involves storing samples of a signal from a **first** excitation signal generator. In response to a signal indicating bit erasure, a **second** excitation signal is synthesised based on previously stored samples of the **first** excitation signal. The **second** signal is filtered to synthesise a human type speech signal...

...Abstract (Equivalent): use by a decoder which experiences an erasure of input bits, the decoder including a **first** excitation signal generator responsive to said input bits and a synthesis filter responsive to an ...

...storing samples of a **first** excitation signal generated by said **first** excitation signal generator...

...responsive to a signal indicating the erasure of input bits, synthesizing a **second** excitation signal based on previously stored

samples of the **first** excitation signal; and...

...filtering said **second** excitation signal to synthesize said signal reflecting human speech...

...wherein the step of synthesizing a **second** excitation signal comprises the steps of...

...forming said **second** excitation signal based on said identified set of excitation signal samples...

...Title Terms: **FRAME** ;

International Patent Class (Main): **G10L-003/02** ...

... **G10L-005/02** ...

... **G10L-009/00** ...

... **G10L-009/14** ...

... **G10L-019/00** ...

... **G10L-019/04**

International Patent Class (Additional): **G10L-009/18** ...

... **G10L-019/12**

21/3,K/4 (Item 4 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2003 Thomson Derwent. All rts. reserv.

009997939 **Image available**
WPI Acc No: 1994-265650/199433
XRPX Acc No: N94-209082

Block size determination method for transform coder - defining audio signals into time intervals according to temporal masking properties of human auditory system and selecting block size by comparing differences between peak values in each time interval

Patent Assignee: MATSUSHITA ELECTRIC IND CO LTD (MATU); MATSUSHITA ELEC IND CO LTD (MATU); MATSUSHITA DENKI SANGYO KK (MATU)

Inventor: TEH D H

Number of Countries: 005 Number of Patents: 006

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
EP 612158	A1	19940824	EP 94101182	A	19940127	199433 B
JP 6242797	A	19940902	JP 9330711	A	19930219	199440
US 5651089	A	19970722	US 93161797	A	19931206	199735
EP 612158	B1	20000405	EP 94101182	A	19940127	200021
DE 69423803	E	20000511	DE 623803	A	19940127	200030
			EP 94101182	A	19940127	
JP 3088580	B2	20000918	JP 9330711	A	19930219	200048

Priority Applications (No Type Date): JP 9330711 A 19930219

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
EP 612158	A1	E	10	H04B-001/66	

Designated States (Regional): DE FR GB

JP 6242797	A	8	G10L-009/18
------------	---	---	-------------

US 5651089	A	10	G10L-003/02
------------	---	----	-------------

EP 612158	B1	E	H04B-001/66
-----------	----	---	-------------

Designated States (Regional): DE FR GB

DE 69423803 E H04B-001/66 Based on patent EP 612158
JP 3088580 B2 8 G10L-019/02 Previous Publ. patent JP 6242797

... temporal masking properties of human auditory system and selecting block size by comparing differences between peak values in each time interval

...Abstract (Basic): process in which digital audio signals having spectral and temporal structure are decomposed into several frames, involves defining the audio signal into time intervals according to a temporal masking properties of the human auditory system, and obtaining a peak value in each of the time intervals...

...Differences among the peak values are calculated, and a block size is selected based upon a comparison of the...

...Abstract (Equivalent): A method of determining a block size for a frame which is a part of a digital audio signal having temporal structure, said block size being used for a transform coding process which composes digital audio signals into frequency spectral frames, comprising steps of...

...said frame being composed of four continuous time intervals...

...obtaining a peak value in each of said time intervals...

...calculating a first difference between said peak value of one of said time intervals and said peak value of an adjacent time interval ...

...calculating a second difference between said peak value of said one of said time intervals and said peak value of another of said time intervals...

...selecting said block size for said frame, based on whether said first difference or said second difference exceeds a predefined value...

...Title Terms: PEAK ;

International Patent Class (Main): G10L-003/02 ...

... G10L-009/18 ...

... G10L-019/02 .

International Patent Class (Additional): G10L-019/00 ...

21/3,K/5 (Item 5 from file: 350)

DIALOG(R)File 350:Derwent WPIX

(c) 2003 Thomson Derwent. All rts. reserv.

004132086

WPI Acc No: 1984-277626/198445

XRXPX Acc No: N84-207236

Channel synthesis subassembly for vocoder - has compensation for parasitic speech modulation by fast-response analysis chain correcting gain of modulator

Patent Assignee: ZURCHER J F (ZURC-I)

Inventor: ZURCHER J F

Number of Countries: 005 Number of Patents: 004

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
EP 124411	A	19841107	EP 84400744	A	19840413	198445 B
FR 2544901	A	19841026				198449

EP 124411 B 19871119
DE 3467650 G 19771223

198746
198801

Priority Applications (No Type Date): FR 836471 A 19830420

Patent Details:

Patent No	Kind	Land	Pg	Main IPC	Filing Notes
-----------	------	------	----	----------	--------------

EP 124411	A	F	29		
-----------	---	---	----	--	--

Designated States (Regional): DE GB IT NL

EP 124411	B	F		
-----------	---	---	--	--

Designated States (Regional): DE GB IT NL

...Abstract (Basic): 48 sub 1 -48 sub n), rectifier (50) and low-pass filter (52) or a **peak** detector following more rapidly the possible abrupt increases in frequency-localised energy...

...Abstract (Equivalent): channel vocoder including a means (36) of reception and extraction of the data in a **frame**, a means (38) of producing an excitation signal, the said synthesis subassembly including: n synthesis...

...means (38) of producing the excitation signal, at least one subtractor (62) receiving on a **first** input the energy signal delivered by a synthesis channel and on a **second** input the signal coming from the means of reception (36) and representing the energy signal...

International Patent Class (Additional): **G10L-001/06** ...

... **G10L-007/04**

?

23/3,K/1 (Item 1 from file: 347)
DIALOG(R)File 347:JAPIO
(c) 2003 JPO & JAPIO. All rts. reserv.

06513376 **Image available**
ACOUSTIC SIGNAL ENCODING METHOD

PUB. NO.: 2000-099093 [JP 2000099093 A]
PUBLISHED: April 07, 2000 (20000407)
INVENTOR(s): MOTEKI TOSHIO
APPLICANT(s): DAINIPPON PRINTING CO LTD
APPL. NO.: 10-283454 [JP 98283454]
FILED: September 18, 1998 (19980918)

ABSTRACT

... acoustic signal to be encoded is PCM(pulse code modulation)-encoded and taken in as **acoustic data**, and plural unit **sections** are set on a time base. Fourier conversion is performed for each unit **section** and a spectrum **S** is obtained. The prescribed threshold value **L** is set, a series of continuous **part** of which the intensity is this threshold value **L** or more out of the spectrum **S** is recognized as formant **F1-F5** being peculiar in vocal respectively. The maximum **peak** frequency in each formant is extracted as a representative frequency representing the formant, and one ...
... Each representative frequency is replaced by a note number of MIDI(musical instrument.digital interface) **data**, an **acoustic** signal in the unit **section** is encoded by this note number.

COPYRIGHT: (C)2000,JPO

23/3,K/2 (Item 2 from file: 347)
DIALOG(R)File 347:JAPIO
(c) 2003 JPO & JAPIO. All rts. reserv.

05938345 **Image available**
OCEANOGRAPHIC ACOUSTIC TOMOGRAPHIC DATA ANALYZER

PUB. NO.: 10-221445 [JP 10221445 A]
PUBLISHED: August 21, 1998 (19980821)
INVENTOR(s): ARAYA TOMIO
APPLICANT(s): OKI ELECTRIC IND CO LTD [000029] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 09-021416 [JP 9721416]
FILED: February 04, 1997 (19970204)

OCEANOGRAPHIC ACOUSTIC TOMOGRAPHIC DATA ANALYZER

ABSTRACT

PROBLEM TO BE SOLVED: To track the **peak** of received signal data by eliminating the influence of tidal phenomena by sampling received signal...

...a receiving system signal data memory 1 at every observation from a data input-output **section** 11. The received signal data are **divided** into a **plurality** of data groups by sampling the data in a period which is close to the tidal periodicity and tracking is performed in accordance with the received signal data in each **divided** data group. Of the periodically observed received signal data, namely, a **plurality** of observed received signal data observed in a period which is close to the lunar...
?

File 2:INSPEC 1969-2003/Nov W4
 (c) 2003 Institution of Electrical Engineers
 File 6:NTIS 1964-2003/Nov W5
 (c) 2003 NTIS, Intl Cpyrht All Rights Res
 File 8:Ei Compendex(R) 1970-2003/Nov W4
 (c) 2003 Elsevier Eng. Info. Inc.
 File 34:SciSearch(R) Cited Ref Sci 1990-2003/Nov W4
 (c) 2003 Inst for Sci Info
 File 35:Dissertation Abs Online 1861-2003/Oct
 (c) 2003 ProQuest Info&Learning
 File 65:Inside Conferences 1993-2003/Nov W5
 (c) 2003 BLDSC all rts. reserv.
 File 94:JICST-EPlus 1985-2003/Nov W5
 (c) 2003 Japan Science and Tech Corp (JST)
 File 95:TEME-Technology & Management 1989-2003/Nov W3
 (c) 2003 FIZ TECHNIK
 File 99:Wilson Appl. Sci & Tech Abs 1983-2003/Oct
 (c) 2003 The HW Wilson Co.
 File 144:Pascal 1973-2003/Nov W4
 (c) 2003 INIST/CNRS
 File 233:Internet & Personal Comp. Abs. 1981-2003/Jul
 (c) 2003, EBSCO Pub.
 File 239:Mathsci 1940-2003/Jan
 (c) 2003 American Mathematical Society
 File 434:SciSearch(R) Cited Ref Sci 1974-1989/Dec
 (c) 1998 Inst for Sci Info
 File 583:Gale Group Globalbase(TM) 1986-2002/Dec 13
 (c) 2002 The Gale Group
 File 603:Newspaper Abstracts 1984-1988
 (c) 2001 ProQuest Info&Learning
 File 483:Newspaper Abs Daily 1986-2003/Dec 01
 (c) 2003 ProQuest Info&Learning
 ? ds

Set	Items	Description
S1	1214	CEPSTRAL(3N)COEFFICIENT?
S2	653	(COMPAR? OR EVALUAT? OR DETERMIN? OR ASSESS? OR DISCERN? OR DISTINGUISH?) AND S1
S3	902098	PEAK?
S4	132825	(MANY OR MULTIPLE OR SEVERAL OR MULTI OR NUMEROUS OR PLURAL?) AND S3
S5	370130	FRAME??
S6	60269	SPEECH(3N)RECOG?
S7	2149	ACOUSTIC(3N)DATA AND (PARTITION? OR DIVID? OR SEPARAT? OR - DIVISION? OR PART OR PARTS OR SECTION?? OR SEGMENT?? OR PORTION?? OR FRAGMENT? OR PIECES OR SECTOR??)
S8	3113	AU=(SHU, C? OR SHU, H? OR SHU C? OR SHU H?)
S9	0	S2 AND S4 AND S5
S10	0	S8 AND S1
S11	6	S2 AND S4
S12	2	RD S11 (unique items)
S13	9	S1 AND S4
S14	3	S13 NOT S11
S15	2	RD S14 (unique items)
S16	29	CEPSTRAL AND PEAK? AND FRAME?
S17	0	S16 AND S7
S18	3	S16 AND SEGMENT?
S19	3	S18 NOT (S13 OR S11)
S20	3	RD S19 (unique items)
S21	10	S8 AND SPEECH
S22	0	S21 AND CEPSTRAL

S23 0 S21 AND PEAK??
S24 10 S21 NOT (S18 OR S13 OR S11)
S25 7 RD S24 (unique items)

12/3,K/1 (Item 1 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2003 Institution of Electrical Engineers. All rts. reserv.

5689535 INSPEC Abstract Number: B9710-6130-481, C9710-1250C-137

Title: A model of dynamic auditory perception and its application to robust word recognition

Author(s): Strope, B.; Alwan, A.

Author Affiliation: Dept. of Electr. Eng., California Univ., Los Angeles, CA, USA

Journal: IEEE Transactions on Speech and Audio Processing vol.5, no.5 p.451-64

Publisher: IEEE,

Publication Date: Sept. 1997 Country of Publication: USA

CODEN: IESPEJ ISSN: 1063-6676

SICI: 1063-6676(199709)5:5L.451:MDAP;1-W

Material Identity Number: P947-97005

U.S. Copyright Clearance Center Code: 1063-6676/97/\$10.00

Language: English

Subfile: B C

Copyright 1997, IEE

...Abstract: common automatic speech recognition (ASR) front end and provide adaptation and isolation of local spectral **peaks**. A dynamic model consisting of a linear filterbank with a novel additive logarithmic adaptation stage...

... An extensive series of perceptual forward masking experiments, together with previously reported forward masking data, **determine** the model's dynamic parameters. Once parameterized, the simple exponential dynamic mechanism predicts the nature of forward masking data from **several** studies across wide ranging frequencies, input levels, and probe delay times. An initial **evaluation** of the dynamic model together with a local **peak** isolation mechanism as a front end for dynamic time warp (DTW) and hidden Markov model (HMM) word recognition systems shows an improvement in robustness to background noise when. **compared** to Mel-frequency **cepstral coefficients** (MFCC), linear prediction **cepstral coefficients** (LPCC), and relative spectra (RASTA) based front ends.

...Identifiers: local spectral **peaks** isolation...

...local spectral **peaks** ; ...

...Mel-frequency **cepstral coefficients** ; ...

...linear prediction **cepstral coefficients** ;

12/3,K/2 (Item 2 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2003 Institution of Electrical Engineers. All rts. reserv.

03853986 INSPEC Abstract Number: B91026300

Title: Towards feature-based speech metric

Author(s): Bayya, A.; Hermansky, H.

Author Affiliation: US West Adv. Technol., Englewood, CO, USA

Conference Title: ICASSP 90. 1990 International Conference on Acoustics, Speech and Signal Processing (Cat. No.90CH2847-2) p.781-4 vol.2

Publisher: IEEE, New York, NY, USA

Publication Date: 1990 Country of Publication: USA 5 vol. 2970 pp.

U.S. Copyright Clearance Center Code: CH2847-2/90/0000-0781\$01.00

Conference Sponsor: IEEE
Conference Date: 3-6 April 1990 Conference Location: Albuquerque, NM,
USA
Language: English
Subfile: B

Abstract: A speech metric which directly uses spectral features such as spectral **peak** frequencies and bandwidths is proposed and **evaluated**. The spectral features either are derived directly by solving the all-pole model polynomial to get spectral **peak** frequencies and bandwidths and fitting the linear regression line to the logarithmic spectrum of the model or are estimated as a linear combination of the **several** lower **cepstral coefficients** of the all-pole model spectrum. The performance of the studied metric is speaker-independent...

Identifiers: **peak** bandwidths...

...spectral **peak** frequencies...

?

15/3,K/1 (Item 1 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2003 Institution of Electrical Engineers. All rts. reserv.

5472373 INSPEC Abstract Number: B9702-6130-233, C9702-1250C-071

Title: An approach to speaker adaptation based on analytic functions

Author(s): McDonough, J.; Zavaliagkos, G.; Gish, H.

Author Affiliation: BBN Syst. & Technol. Corp., Cambridge, MA, USA

Conference Title: 1996 IEEE International Conference on Acoustics, Speech, and Signal Processing Conference Proceedings (Cat. No.96CH35903)

Part vol. 2 p.721-4 vol. 2

Publisher: IEEE, New York, NY, USA

Publication Date: 1996 Country of Publication: USA 6 vol. lvii+3588

pp.

ISBN: 0 7803 3192 3 Material Identity Number: XX96-02717

U.S. Copyright Clearance Center Code: 0 7803 3192 3/96/\$5.00

Conference Title: 1996 IEEE International Conference on Acoustics, Speech, and Signal Processing Conference Proceedings

Conference Sponsor: Signal Process. Soc. IEEE

Conference Date: 7-10 May 1996 Conference Location: Atlanta, GA, USA

Language: English

Subfile: B C

Copyright 1997, IEE

...Abstract: formulate a novel approach to speaker adaptation. It is predicated upon the fact that the **cepstral coefficients** used as feature vectors in most state of the art speech recognition systems are coefficients...

... an analytic function of a complex-valued argument. This analytic function can be characterized by several poles and zeros in the complex plane corresponding to spectral **peaks** and nulls, respectively. Speaker adaptation can be viewed as the estimation of a function that...

...Identifiers: **cepstral coefficients** ; ...

...spectral **peaks** ;

15/3,K/2 (Item 1 from file: 35)

DIALOG(R)File 35:Dissertation Abs Online

(c) 2003 ProQuest Info&Learning. All rts. reserv.

01797549 ORDER NO: AADAA-I9935248

THE RATIO SPECTRUM (DISCRETE TIME, POWER SPECTRUM REPRESENTATION)

Author: LIM, SHAO-JEN

Degree: PH.D.

Year: 1999

Corporate Source/Institution: UNIVERSITY OF FLORIDA (0070)

Source: VOLUME 60/06-B OF DISSERTATION ABSTRACTS INTERNATIONAL.

PAGE 2858. 131 PAGES

...have developed a novel power spectrum representation called the *ratio spectrum* that provides many advantages over the standard power spectrum, particularly for analog implementations. The ratio spectrum is formed...

...vector for speech recognition systems. Initial results show that the ratio spectrum outperforms LP-derived **cepstral coefficients** , **peaks** found through LP analysis and filter banks in phoneme recognition experiments. (4) The ratio...

20/3,K/1 (Item 1 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2003 Institution of Electrical Engineers. All rts. reserv.

6754838 INSPEC Abstract Number: B2000-12-6130E-158, C2000-12-1250C-116

Title: On the use of variable frame rate analysis in speech recognition

Author(s): Qifeng Zhu; Alwan, A.

Author Affiliation: Dept. of Electr. Eng., California Univ., Los Angeles, CA, USA

Conference Title: 2000 IEEE International Conference on Acoustics, Speech, and Signal Processing. Proceedings (Cat. No.00CH37100) Part vol.3 p.1783-6 vol.3

Publisher: IEEE, Piscataway, NJ, USA

Publication Date: 2000 Country of Publication: USA 6 vol. 1xxx+3906 pp.

ISBN: 0 7803 6293 4 Material Identity Number: XX-2000-01776

U.S. Copyright Clearance Center Code: 0 7803 6293 4/2000/\$10.00

Conference Title: Proceedings of 2000 International Conference on Acoustics, Speech and Signal Processing

Conference Sponsor: IEEE; Signal Process. Soc

Conference Date: 5-9 June 2000 Conference Location: Istanbul, Turkey

Language: English

Subfile: B C

Copyright 2000, IEE

Title: On the use of variable frame rate analysis in speech recognition

...Abstract: discriminating and identifying speech sounds. These changes can occur over very short time intervals. Computing **frames** every 10 ms, as commonly done in recognition systems, is not sufficient to capture such dynamic changes. In this paper, we propose a variable **frame** rate (VFR) algorithm. The algorithm results in an increased number of **frames** for rapidly-changing **segments** with relatively high energy and less **frames** for steady-state **segments**. The current implementation used an average data rate which is less than 100 **frames** per second. For an isolated word recognition task, and using an HMM-based speech recognition...

... accuracy especially at low signal-to-noise ratios. The technique was evaluated with mel frequency **cepstral** coefficient (MFCC) vectors and MFCC vectors with enhanced **peak** isolation.

Descriptors: **cepstral** analysis...

Identifiers: variable **frame** rate analysis...

...rapidly-changing **segments** ; ...

...steady-state **segments** ; ...

...mel frequency **cepstral** coefficient

20/3,K/2 (Item 2 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2003 Institution of Electrical Engineers. All rts. reserv.

5362263 INSPEC Abstract Number: B9610-6130-052, C9610-1250C-018

Title: Feature extraction for three- segmental speech signals

Author(s): Tung, S.-L.; Juang, Y.-T.

Author Affiliation: Dept. of Electr. Eng., Nat. Central Univ., Chung-Li, Taiwan

Conference Title: 1995 International Symposium on Communications Part vol.1 p.287-94 vol.1

Publisher: Nat. Taiwan Univ, Taipei, Taiwan

Publication Date: 1995 Country of Publication: Taiwan 2 vol.
xxii+1235 pp.
Material Identity Number: XX95-01599
Conference Title: Proceedings of 1995 International Symposium on
Communications. ISCOM'95
Conference Sponsor: Ministr. Educ.; Nat. Sci. Council; Ind. Technol.; et
al
Conference Date: 27-29 Dec. 1995 Conference Location: Taipei, Taiwan
Language: English
Subfile: B C
Copyright 1996, IEE

Title: Feature extraction for three- segmental speech signals

Abstract: Generally, the feature extraction of speech signals is to obtain the LPC cepstrum **frame** by **frame**, but it requires a lot of memories to represent each word and does not consider...

... not less than the others. Firstly, we propose a simple way to find the pitch **peaks**, then according to the pitch **peaks**, we divide each word into three **segments** : consonant- **segment**, vowel- **segment** and residual- **segment**, finally we select one **frame** in each **segment** to compute the LPC-cepstrum. From our experiments, this method obtains a good result and shows that the variances between **cepstral** coefficients are smaller than the others.

...Identifiers: three- **segmental** speech signals...

...pitch **peaks** ; ...

...consonant- **segment** ; ...

...vowel- **segment** ; ...

...residual- **segment**

20/3,K/3 (Item 1 from file: 144)

DIALOG(R)File 144:Pascal
(c) 2003 INIST/CNRS. All rts. reserv.

15090562 PASCAL No.: 01-0250265
Efficient automatic recognition of spoken digit strings
OSHAUGHNESSY Douglas; TOLBA Hesham
INRS-Telecommunications, 900 de la Gauchetiere west, P.O. Box 644,
Montreal, PQ H5A 1C6, Canada
Journal: The Journal of the Acoustical Society of America, 2001-05-01,
109 (5) p. 2316
Language: English

Copyright (c) 2001 American Institute of Physics. All rights reserved.

... application, such recognition was investigated here under different conditions. Traditional hidden Markov model approaches with **cepstral** analysis were not used, because they are computationally intensive and have not always worked well under adverse acoustic conditions. Simpler spectral analysis was used, combined with a **segmental** approach. The analysis focuses on locations of spectral **peaks**, similar to formant tracking, but without the need to estimate **peaks** for all time **frames**. The limited nature of the vocabulary (i.e., ten digits) allows this simpler approach. High...
?

25/3,K/1 (Item 1 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2003 Institution of Electrical Engineers. All rts. reserv.

7725113 INSPEC Abstract Number: B2003-10-6135E-073, C2003-10-5260B-263

Title: Segmentation of color lip images by spatial fuzzy clustering

Author(s): Liew, A.W.-C.; Shu Hung Leung ; Wing Hong Lau

Author Affiliation: Dept. of Comput. Eng. & Inf. Technol., City Univ. of Hong Kong, China

Journal: IEEE Transactions on Fuzzy Systems vol.11, no.4 p.542-9

Publisher: IEEE,

Publication Date: Aug. 2003 Country of Publication: USA

CODEN: IEFSEV ISSN: 1063-6706

SICI: 1063-6706(200308)11:4L.542:SCIS;1-T

Material Identity Number: P984-2003-004

U.S. Copyright Clearance Center Code: 1063-6706/03/\$17.00

Language: English

Subfile: B C

Copyright 2003, IEE

Author(s): Liew, A.W.-C.; Shu Hung Leung ; Wing Hong Lau

...Descriptors: speech recognition

...Identifiers: automatic speech recognition system

25/3,K/2 (Item 2 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2003 Institution of Electrical Engineers. All rts. reserv.

7500355 INSPEC Abstract Number: B2003-02-6135E-050, C2003-02-1250M-032

Title: Lip contour extraction from color images using a deformable model

Author(s): Liew, A.W.-C.; Shu Hung Leung ; Wing Hong Lau

Author Affiliation: Dept. of Comput. Eng. & Inf. Technol., City Univ. of Hong Kong, Kowloon, China

Journal: Pattern Recognition vol.35, no.12 p.2949-62

Publisher: Elsevier,

Publication Date: Dec. 2002 Country of Publication: UK

CODEN: PTNRA8 ISSN: 0031-3203

SICI: 0031-3203(200212)35:12L.2949:CEFC;1-W

Material Identity Number: P133-2002-010

U.S. Copyright Clearance Center Code: 0031-3203/02/\$22.00

Language: English

Subfile: B C

Copyright 2003, IEE

Author(s): Liew, A.W.-C.; Shu Hung Leung ; Wing Hong Lau

...Abstract: use of visual information from lip movements can improve the accuracy and robustness of a speech recognition system. In this paper, a region-based lip contour extraction algorithm based on deformable...

...Descriptors: speech recognition

...Identifiers: speech recognition system

25/3,K/3 (Item 1 from file: 8)

DIALOG(R)File 8:Ei Compendex(R)

(c) 2003 Elsevier Eng. Info. Inc. All rts. reserv.

05392755 E.I. No: E2099104846999

Title: Selected phoneme rejection grammar for a speech recognition system

Author: **Shu, Chang-Qing**
Corporate Source: BBN Technologies, Cambridge, MA, USA
Conference Title: Proceedings of the 1998 4th International Conference on
Signal Processing Proceedings, ICSP '98
Conference Location: Beijing, China Conference Date: 19981012-19981016
E.I. Conference No.: 55222
Source: International Conference on Signal Processing Proceedings, ICSP v
1 1998. p 646-649
Publication Year: 1998
CODEN: 002534
Language: English

Title: Selected phoneme rejection grammar for a speech recognition system

Author: **Shu, Chang-Qing**
Descriptors: **Speech recognition; Pattern recognition systems; Computational grammars**

25/3,K/4 (Item 2 from file: 8)

DIALOG(R)File 8:EI Compendex(R)
(c) 2003 Elsevier Eng. Info. Inc. All rts. reserv.

04251323 E.I. No: EIP95092853449

Title: Duration modeling in large vocabulary speech recognition
Author: Anastasakos, Anastasios; Schwartz, Richard; **Shu, Han**
Corporate Source: Northeastern Univ, Boston, MA, USA
Conference Title: Proceedings of the 1995 International Conference on
Acoustics, Speech, and Signal Processing. Part 1 (of 5)
Conference Location: Detroit, MI, USA Conference Date:
19950509-19950512
E.I. Conference No.: 43559
Source: Speech ICASSP, IEEE International Conference on Acoustics, Speech
and Signal Processing - Proceedings v 1 1995. IEEE, Piscataway, NJ,
USA, 95CH35732. p 628-631
Publication Year: 1995
CODEN: IPRODJ ISSN: 0736-7791
Language: English

Title: Duration modeling in large vocabulary speech recognition

Author: Anastasakos, Anastasios; Schwartz, Richard; **Shu, Han**
Abstract: This paper presents a study of different methods for phoneme
duration modeling in large vocabulary **speech** recognition. We investigate
the employment of phoneme duration and the effect of context, speaking rate
and lexical stress in the duration of phoneme segments in a large
vocabulary **speech** recognition system. The duration models are used in a
postprocessing phase of BYBLOS, our baseline...

Descriptors: **Speech recognition; Character recognition; Markov
processes; Mathematical models; Speech analysis; Algorithms; Probability**

Identifiers: **Vocabulary speech recognition system; Hidden semi-Markov
models; Parzen-window method; Duration modeling; Human listeners**

25/3,K/5 (Item 1 from file: 144)

DIALOG(R)File 144:Pascal
(c) 2003 INIST/CNRS. All rts. reserv.

14740045 PASCAL No.: 00-0417150

Rejection grammar using selected phonemes for speech recognition system
SHU Chang-Qing

Journal: The Journal of the Acoustical Society of America, 2000-10, 108
(4) p. 1383
Language: English

Copyright (c) 2000 American Institute of Physics. All rights reserved.

Rejection grammar using selected phonemes for speech recognition system
SHU Chang-Qing

English Descriptors: Instrumentation; Measuring methods; **Speech**
processing; **Speech** recognition equipment; Natural languages; **Speech**
intelligibility

25/3,K/6 (Item 2 from file: 144)

DIALOG(R)File 144:Pascal
(c) 2003 INIST/CNRS. All rts. reserv.

13451871 PASCAL No.: 98-0147032
Total least squares linear prediction for frequency estimation with
frequency Weighting
ICASSP 97 : international conference on acoustics, speech , and signal
processing : Munich, April 21-24, 1997. Volume V: Statistical signal and
array processing, applications

SHU HUNG LEUNG ; TIN HO LEE; WING HONG LAU
Department of Electronic Engineering, City University of Hong Kong, 83
Tat Chee Avenue, Kowloon, Hong Kong
IEEE, New York NY, United States.
International conference on acoustics, speech, and signal processing (
Munich DEU) 1997-04-21
1997 3993-3996
Publisher: IEEE Computer Society Press, Washington DC
Language: English

Copyright (c) 1998 INIST-CNRS. All rights reserved.

ICASSP 97 : international conference on acoustics, speech , and signal
processing : Munich, April 21-24, 1997. Volume V: Statistical signal and
array processing...

SHU HUNG LEUNG ; TIN HO LEE; WING HONG LAU

25/3,K/7 (Item 1 from file: 239)

DIALOG(R)File 239:Mathsci
(c) 2003 American Mathematical Society. All rts. reserv.

03357631 CMP 1 927 494
A new method for fast computing Legendre moments.
Dong, Jian (Biomedical Engineering Department, Southeast University,
Nanjing 210018, Jiangsu, Peoples Republic of China)
Zhou, Jin Dan (Biomedical Engineering Department, Southeast University,
Nanjing 210018, Jiangsu, Peoples Republic of China)
Zhou, Fei Ya (Biomedical Engineering Department, Southeast University,
Nanjing 210018, Jiangsu, Peoples Republic of China)
Shu, Hua Zhong (Biomedical Engineering Department, Southeast
University, Nanjing 210018, Jiangsu, Peoples Republic of China)
Haigron, Pascal (Laboratoire Antennes et Traitement du Signal, Universite
de Rennes I, 35042 Rennes, France)
Luo, Li Min (Biomedical Engineering Department, Southeast University,
Nanjing 210018, Jiangsu, Peoples Republic of China)

Corporate Source Codes: PRC-SEU-BM; PRC-SEU-BM; PRC-SEU-BM; PRC-SEU-BM;
F-RENN-ANT; PRC-SEU-BM

Chinese J. Comput.

Chinese Journal of Computers. Jisuanji Xuebao, 2002, 25, no. 6,
576--581. ISSN: 0254-4164 CODEN: JIXUDT

Language: Chinese Summary Language: English, Chinese
Subfile: CMP (Current Mathematical Publications) AMS
...Nanjing 210018, Jiangsu, Peoples Republic of China)

Shu, Hua Zhong ...

Descriptors: ...; in a specific mathematical area, see Section --04 in
that area)-Artificial intelligence-Pattern recognition, speech
recognition (For cluster analysis, see 62H30...
?

File 348:EUROPEAN PATENTS 1978-2003/Nov W04

(c) 2003 European Patent Office

File 349:PCT FULLTEXT 1979-2002/UB=20031127,UT=20031120

(c) 2003 WIPO/Univentio

? ds

Set	Items	Description
S1	323	CEPSTRAL(3N) (COEFFICIENT? OR PARAMETER? OR VALUES)
S2	49	(COMPAR? OR EVALUAT? OR DETERMIN? OR ASSESS? OR DISCERN? OR DISTINGUISH?) (5N)S1
S3	149105	PEAK?
S4	6587	(MANY OR MULTIPLE OR SEVERAL OR MULTI OR NUMEROUS OR PLURAL?) (5N)S3
S5	257336	FRAME??
S6	5863	SPEECH(3N)RECOG?
S7	371	ACOUSTIC(3N)DATA(10N) (PARTITION? OR DIVID? OR SEPARAT? OR - DIVISION? OR PART OR PARTS OR SECTION?? OR SEGMENT?? OR PORTION?? OR FRAGMENT? OR PIECES OR SECTOR??)
S8	6444	IC=G10L?
S9	2	S2(S)S3
S10	19	S1(S)FIRST(S)SECOND(S)S5
S11	1	S10(S)S3
S12	0	S11 NOT S9
S13	17	S10 AND S8
S14	16	S13 NOT S9
S15	1	S7(S)FIRST(S)SECOND(10N)S3
S16	1	S15 NOT (S13 OR S9)
S17	0	S16 NOT ULTRSONIC
S18	2	S4(S)S5(S)S6
S19	2	S18 NOT (S15 OR S13 OR S9)
S20	205	(COMPAR? OR EVALUAT? OR DETERMIN? OR ASSESS? OR DISCERN? OR DISTINGUISH?) (3N)S5(7N)S3
S21	0	S20(S)S7
S22	47	S20 AND S8
S23	0	S22(S)CEPSTRAL
S24	2	S22(S)S6

9/3,K/1 (Item 1 from file: 349)

DIALOG(R)File 349:PCT FULLTEXT

(c) 2003 WIPO/Univentio. All rts. reserv.

00799959 **Image available**

METHODS AND APPARATUSES FOR SIGNAL ANALYSIS

PROCEDES ET APPAREILS D'ANALYSE DE SIGNAUX

Patent Applicant/Assignee:

HUQ SPEECH TECHNOLOGIES B V, Zilverlaan 2, NL-9743 RK Groningen, NL, NL
(Residence), NL (Nationality), (For all designated states except: US)

Patent Applicant/Inventor:

ANDRINGA Tjeerd Catharinus, Paterswoldseweg 324, NL-9727 BX Groningen, NL
, NL (Residence), NL (Nationality), (Designated only for: US)

DUIFHUIS Hendrikus, Zonnebloemweg 21, NL-9765 HW Paterswolde, NL, NL
(Residence), NL (Nationality), (Designated only for: US)

VAN HENGEL Pieter Willem Jan, Hiddemaheerd 38, NL-9737 JP Groningen, NL
NL (Residence), NL (Nationality), (Designated only for: US)

HEEMSKERK Michael Gerardus, Aquamarijnstraat 421, NL-9743 PK Groningen,
NL, NL (Residence), NL (Nationality), (Designated only for: US)

NILLESEN Maartje Marjolein, Friesestraatweg 5A, NL-9718 NA Groningen, NL,
NL (Residence), NL (Nationality), (Designated only for: US)

Legal Representative:

PRINS Ir A W (agent), Vereenigde, Nieuwe Parklaan 97, NL-2587 BN The
Hague, NL,

Patent and Priority Information (Country, Number, Date):

Patent: WO 200133547 A1 20010510 (WO 0133547)

Application: WO 2000NL808 20001106 (PCT/WO NL0000808)

Priority Application: NL 1013500 19991105

Designated States: AE AG AL AM AT AU AZ BA BB BG BR BY BZ CA CH CN CR CU CZ
DE DK DM DZ EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ
LC LK LR LS LT LU LV MA MD MG MK MN MW MX MZ NO NZ PL PT RO RU SD SE SG
SI SK SL TJ TM TR TT TZ UA UG US UZ VN YU ZA ZW
(EP) AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE TR
(OA) BF BJ CF CG CI CM GA GN GW ML MR NE SN TD TG
(AP) GH GM KE LS MW MZ SD SL SZ TZ UG ZW
(EA) AM AZ BY KG KZ MD RU TJ TM

Publication Language: English

Filing Language: English

Fulltext Word Count: 24777

Fulltext Availability:

Claims

Claim

... mask is often a subset of the CPC mask and/or the
TAC mask. The **peaks** of the TAC-selection show up in the CPC mask as
well. The information represented...The ridges are formed, as in the fun
damental period estimation algorithm, by combining successive **peaks**
that
differ less than 2 in terms of segment number. Ridges longer than 15 ins
...

...This stage leads to high- frequency contribution
with unrealistic upward and downward slopes. The black **peaks** in the
upper panel of figure 2.22 show this clearly. To make the reconstruction
...

...that represent the masking effects consistent with a source that excites
the position of the **peak** next to the flank. These can again be
estimated from the sine-responses and added...be followed by an algorithm
that estimates the spectral envelope, e.g. by connecting the **peaks** of

consecutive harmonics. As a final step, the envelope of the cochleogram must be coded...

...are enhanced by values between 1 and 5, the spectral envelope is coded with 12 **cepstral coefficients**. Compared to the lower panel of figure 2.24, the high-frequency segments are much more...

9/3,K/2 (Item 2 from file: 349)
DIALOG(R)File 349:PCT FULLTEXT
(c) 2003 WIPO/Univentio. All rts. reserv.

00155778 **Image available**

SPEECH ANALYSIS

ANALYSE DE LA PAROLE

Patent Applicant/Assignee:

THE UNIVERSITY OF MELBOURNE,
JOHNSON Donald Archibald Harvey,
NANDAGOPAL Daraisamy,

Inventor(s):

JOHNSON Donald Archibald Harvey,
NANDAGOPAL Daraisamy,

Patent and Priority Information (Country, Number, Date):

Patent: WO 8902145 A1 19890309

Application: WO 88AU325 19880825 (PCT/WO AU8800325)

Priority Application: AU 873993 19870825

Designated States: AT AU BE CH DE FR GB IT JP LU NL SE US

Publication Language: English

Fulltext Word Count: 2171

Fulltext Availability:

Detailed Description

Detailed Description

... of the

relationship between the negative derivative of phase spectrum (group delay) and the **cepstral coefficients** and hence helps to **evaluate** the group delays from the cep'stral coefficients using the following equation,

co

0 t...

...0'(w) is the negative derivative of phase spectrum

1 5 The positive and negative **peaks** in the negative derivative of phase spectrum correspond to complex poles and complex zeros of...

?

14/3,K/1 (Item 1 from file: 348)
DIALOG(R)File 348:EUROPEAN PATENTS
(c) 2003 European Patent Office. All rts. reserv.

01334214

Spelling speech recognition apparatus

Vorrichtung zur Spracherkennung von buchstabierten Wortern

Appareil de reconnaissance vocale de mots epeles

PATENT ASSIGNEE:

Verbaltek, Inc., (3166380), 2890 Zanker Road, Suite 209, San Jose,
California, (US), (Applicant designated States: all)

INVENTOR:

Pan, James, 94D Escondido Village, Stanford, California 94305, (US)
Kim, Yoon, 1540 Oak Creek Drive, No. 302, Palo Alto, California 94304,
(US)

Chang, Josephine, 1235 Mayberry Lane, San Jose, California 95131, (US)
Chen, Juinn-Yan, 5393 Quebec Common, Fremont, California 94555, (US)

LEGAL REPRESENTATIVE:

Hackney, Nigel John et al (76991), Mewburn Ellis, York House, 23 Kingsway
, London WC2B 6HP, (GB)

PATENT (CC, No, Kind, Date): EP 1139332 A2 011004 (Basic)
EP 1139332 A3 011205
EP 1139332 A9 020320

APPLICATION (CC, No, Date): EP 2000309816 001106;

PRIORITY (CC, No, Date): US 538657 000330

DESIGNATED STATES: AT; BE; CH; CY; DE; DK; ES; FI; FR; GB; GR; IE; IT; LI;
LU; MC; NL; PT; SE; TR

EXTENDED DESIGNATED STATES: AL; LT; LV; MK; RO; SI

INTERNATIONAL PATENT CLASS: G10L-015/00 ; G10L-015/02 ; G10L-015/18

ABSTRACT WORD COUNT: 177

NOTE:

Figure number on first page: 2

LANGUAGE (Publication,Procedural,Application): English; English; English

FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS A	(English)	200140	2026
SPEC A	(English)	200140	6184
Total word count - document A			8210
Total word count - document B			0
Total word count - documents A + B			8210

INTERNATIONAL PATENT CLASS: G10L-015/00 ...

... G10L-015/02 ...

... G10L-015/18

...SPECIFICATION invention. A preemphasizer 301 which preferably is a fixed low-order digital system (typically a **first** -order FIR filter) spectrally flattens the signal $s(n)$, and is described by: where $0.9 \leq a \leq 1.0$. In another embodiment of the invention, preemphasizer 301 is a **first** -order adaptive system having the transfer function where $a(n)$ changes with time (n) according to...

...contemplation of this invention. The zeroth autocorrelation is the frame energy of a given frame. Cepstral coefficient generator 305 converts each frame into cepstral coefficients (the inverse Fourier transform of the log magnitude spectrum, refer below) using Durbin's method, which is known in the art. Tapered cepstral window 306 weights

the cepstral coefficients in order to minimize the effects of noise. Tapered windower 306 is chosen to lower the sensitivity of the low-order cepstral coefficients to overall spectral slope and the high-order cepstral coefficients to noise (or other undesirable variability). Temporal differentiator 307 generates the first time derivative of the cepstral coefficients preferably employing an orthogonal polynomial fit to approximate (in this embodiment, a least-squares estimate...).

...over a finite-length window) to produce processed signal S'(n). In another embodiment, the second time derivative can also be generated by temporal differentiator 307 using approximation techniques known in... extraction which, for example, may produce 50 columns of cepstral coefficients (or vectors) per one second of speech. Letter utterance comparator 201 compares the cepstral distances of each letter with the...

...z utilizing dynamic time warping (DTW). If the inputted speech lasted for two seconds (100 frames), and each portion of the speech file were 25 frames each, letter utterance comparator 201 compares 25 columns of cepstral cepstral vectors with the 26...

...the alphabet in pronunciation database 103. Assuming each letter in pronunciation database 103 is 25 frames long, the DTW comparison is 25 x 25. Because of the vagaries of pronunciation, background...

...an illustration of the distortion scoring technique, in the "s-e-a-t" example, the first letter "s" is initially recognized as "x" so there will be a non-zero distortion...

14/3,K/2 (Item 2 from file: 348)
DIALOG(R)File 348:EUROPEAN PATENTS
(c) 2003 European Patent Office. All rts. reserv.

01331092

Client-server distributed speech recognition
Auf Benutzer/Anbieter verteilte Spracherkennung
Reconnaissance de la parole distribuee en client-serveur
PATENT ASSIGNEE:

Verbaltek, Inc., (3166380), 2890 Zanker Road, Suite 209, San Jose,
California, (US), (Applicant designated States: all)

INVENTOR:

Pan, James, 94D Escondido Village, Stanford, California 94305, (US)

LEGAL REPRESENTATIVE:

Hackney, Nigel John et al (76991), Mewburn Ellis, York House, 23 Kingsway
, London WC2B 6HP, (GB)

PATENT (CC, No, Kind, Date): EP 1136983 A1 010926 (Basic)

APPLICATION (CC, No, Date): EP 2000309800 001106;

PRIORITY (CC, No, Date): US 535431 000323

DESIGNATED STATES: AT; BE; CH; CY; DE; DK; ES; FI; FR; GB; GR; IE; IT; LI;
LU; MC; NL; PT; SE; TR

EXTENDED DESIGNATED STATES: AL; LT; LV; MK; RO; SI

INTERNATIONAL PATENT CLASS: G10L-015/26

ABSTRACT WORD COUNT: 90

NOTE:

Figure number on first page: 2

LANGUAGE (Publication, Procedural, Application): English; English; English
FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS A	(English)	200139	949

SPEC A (English)	200139	6882
Total word count - document A		7831
Total word count - document B		0
Total word count - documents A + B		7831

INTERNATIONAL PATENT CLASS: G10L-015/26

...SPECIFICATION also be advantageously used. The zeroth autocorrelation is the frame energy of a given frame. Cepstral coefficient generator 305 converts each frame into cepstral coefficients (the coefficients of the Fourier transform representation ...using Durbin's method, which is known in the art. Tapered windower 306 weights the cepstral coefficients in order to minimize the effects of noise. Tapered windower 306 is chosen to lower the sensitivity of the low-order cepstral coefficients to overall spectral slope and the high-order cepstral coefficients to noise (or other undesirable variability). Temporal differentiator 307 generates the first time derivative of the cepstral coefficients preferably employing an orthogonal polynomial fit to approximate (in this embodiment, a least-squares estimate...

...over a finite-length window) to produce processed signal S'(n). In another embodiment, the second time derivative can also be generated by temporal differentiator 307 using approximation techniques known in...

14/3,K/3 (Item 3 from file: 348)
 DIALOG(R)File 348:EUROPEAN PATENTS
 (c) 2003 European Patent Office. All rts. reserv.

00711643

Signal bias removal for robust telephone speech recognition.
 Korrektur von Signalverzerrungen fur robuste Spracherkennung ueber Telefon.
 Correction de la distortion du signal pour reconnaissance robuste de la parole transmise par telephone.

PATENT ASSIGNEE:

AT&T Corp., (589370), 32 Avenue of the Americas, New York, NY 10013-2412,
 (US), (applicant designated states: DE;ES;FR;GB;IT)

INVENTOR:

Juang, Biing-Hwang, 8 South Lane, Somerset, New Jersey 07059, (US)
 Rahim, Mazin G., 31 Kimberley Court, Manalapan, New Jersey 07726, (US)

LEGAL REPRESENTATIVE:

Johnston, Kenneth Graham et al (32381), AT&T (UK) Ltd. 5 Mornington Road,
 Woodford Green Essex, IG8 OTU, (GB)

PATENT (CC, No, Kind, Date): EP 674306 A2 950927 (Basic)
 EP 674306 A3 970910

APPLICATION (CC, No, Date): EP 95301747 950316;

PRIORITY (CC, No, Date): US 217035 940324

DESIGNATED STATES: DE; ES; FR; GB; IT

INTERNATIONAL PATENT CLASS: G10L-003/02 ; G10L-003/00

ABSTRACT WORD COUNT: 97

LANGUAGE (Publication,Procedural,Application): English; English; English
 FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS A	(English)	EPAB95	1030
SPEC A	(English)	EPAB95	6424
Total word count - document A			7454
Total word count - document B			0
Total word count - documents A + B			7454

INTERNATIONAL PATENT CLASS: G10L-003/02 ...

... G10L-003/00

...SPECIFICATION speech signal $Y((\omega))$, at step 31, undergoes feature analysis at step 32 to generate **cepstral coefficients**. A **first** index m is set equal to one at step 33, and a **second** index n is also set equal to one at step 34. An estimate of the spectral bias, $/b$, is computed at step 35 for each utterance of T **frames**, such that: (see image in original document)
where the best codeword $z(\text{sub}(t))$ is...
...process is repeated until $n = N$, then step 39 is reached. At step 39 the **second** index m is checked to see if a predetermined number M has been reached. If not, in step 40 the **first** index n is reset to equal one and the **second** index m is incremented by one. Next, vector quantization is performed in step 41 and...

14/3,K/4 (Item 4 from file: 348)

DIALOG(R)File 348:EUROPEAN PATENTS

(c) 2003 European Patent Office. All rts. reserv.

00603097

Keyword/non-keyword classification in isolated word speech recognition
Klassifikation bei Spracherkennung von isolierten Wörtern in
Schlüsselwörter und Nicht-Schlüsselwörter

Classification de mots-cles/de mots non cles dans la reconnaissance du
langage par mots isoles

PATENT ASSIGNEE:

AT&T Corp., (589370), 32 Avenue of the Americas, New York, NY 10013-2412,
(US), (applicant designated states: AT;BE;CH;DE;ES;FR;GB;IT;LI;NL;SE)

INVENTOR:

Sukkar, Rafid Antoon, 68 Forestview Lane, Aurora, Illinois 60504, (US)

LEGAL REPRESENTATIVE:

Watts, Christopher Malcolm Kelway, Dr. et al (37391), Lucent Technologies
(UK) Ltd, 5 Mornington Road, Woodford Green Essex, IG8 0TU, (GB)

PATENT (CC, No, Kind, Date): EP 601778 A1 940615 (Basic)

EP 601778 B1 990310

APPLICATION (CC, No, Date): EP 93309587 931201;

PRIORITY (CC, No, Date): US 989299 921211

DESIGNATED STATES: AT; BE; CH; DE; ES; FR; GB; IT; LI; NL; SE

INTERNATIONAL PATENT CLASS: G10L-005/06 ; G10L-007/08 ; G10L-009/06 ;

G10L-009/18

ABSTRACT WORD COUNT: 151

LANGUAGE (Publication,Procedural,Application): English; English; English
FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS B	(English)	9910	321
CLAIMS B	(German)	9910	331
CLAIMS B	(French)	9910	422
SPEC B	(English)	9910	3464
Total word count - document A			0
Total word count - document B			4538
Total word count - documents A + B			4538

INTERNATIONAL PATENT CLASS: G10L-005/06 ...

... G10L-007/08 ...

... G10L-009/06 ...

... G10L-009/18

...SPECIFICATION basic functionality of the exemplary embodiment of this invention is shown. Digitized speech in 8Hz **frames** is used as input 302 to a linear predictive coding analysis system 304. Linear predictive coding analysis and feature vector generation represent each **frame** with 24 parameters, as shown in Table 1. For purposes of describing this embodiment of...

...length and 15ms update rate. A total of 38 parameters are computed, consisting of the **first** 12 **cepstral coefficients**, their **first** derivatives (called the **Delta Cepstral Coefficients**), the **second** derivatives (called **Delta-Delta Cepstral Coefficients**), Delta energy, and Delta-Delta energy. To reduce the computational load, a subset of 24

...

...Signal Processing, page 501-504, March, 1992. The linear predictive (LP) coding analysis delivers the **first** 10 autocorrelation coefficients to feature vector generator 306. Feature vector generation uses the memory and produces Table 1 on a **frame** by **frame** basis.

Feature vector generator 306 delivers parameters of Table 1 to a Hidden Markov Model...

14/3,K/5 (Item 5 from file: 348)
DIALOG(R)File 348:EUROPEAN PATENTS
(c) 2003 European Patent Office. All rts. reserv.

00421147

Speech recognizer

Einrichtung zur Spracherkennung

Dispositif pour la reconnaissance de la parole

PATENT ASSIGNEE:

MATSUSHITA ELECTRIC INDUSTRIAL CO., LTD., (216883), 1006, Oaza Kadoma, Kadoma-shi, Osaka-fu, 571, (JP), (applicant designated states: DE;FR;GB)

INVENTOR:

Takizawa, Yumi, 30-5-707 Miiminamimachi, Neyagawa-shi, Osaka-fu, (JP)
Hamada, Masahiro, 2-20-1-1427 Higashinakaburi, Hirakata-shi, Osaka-fu, (JP)

LEGAL REPRESENTATIVE:

Eisenfuhr, Speiser & Partner (100151), Martinistraße 24, 28195 Bremen, (DE)

PATENT (CC, No, Kind, Date): EP 421341 A2 910410 (Basic)
EP 421341 A3 920729
EP 421341 B1 990317

APPLICATION (CC, No, Date): EP 90118858 901002;

PRIORITY (CC, No, Date): JP 89259034 891004; JP 90212831 900810

DESIGNATED STATES: DE; FR; GB

INTERNATIONAL PATENT CLASS: G10L-005/06

ABSTRACT WORD COUNT: 133

LANGUAGE (Publication,Procedural,Application): English; English; English

FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS B	(English)	9911	580
CLAIMS B	(German)	9911	488
CLAIMS B	(French)	9911	715

SPEC B	(English)	9911	6759
Total word count - document A		0	
Total word count - document B		8542	
Total word count - documents A + B		8542	

INTERNATIONAL PATENT CLASS: G10L-005/06

...SPECIFICATION 2.5 KHz low-pass filter 18, the reference voice signal is fed to the first analyzer 3. At the first analyzer 3, the LPC cepstral coefficients having a predetermined order is calculated as the recognition parameter. Analysis is performed in the same manner as in the above first and second embodiments. The characteristic parameter in a frame in which the power exceeds a detection threshold value within a predetermined voice interval is...

14/3,K/6 (Item 6 from file: 348)
 DIALOG(R)File 348:EUROPEAN PATENTS
 (c) 2003 European Patent Office. All rts. reserv.

00314930
 Low cost speech recognition system and method.
 Billige Spracherkennungseinrichtung und Verfahren.
 Procede et dispositif economiques pour la reconnaissance de la parole.
 PATENT ASSIGNEE:

TEXAS INSTRUMENTS INCORPORATED, (279070), 13500 North Central Expressway,
 Dallas Texas 75265, (US), (applicant designated states: DE;FR;GB;IT)

INVENTOR:

Doddington, George R., 910 St. Luke Drive, Richardson Texas 75080, (US)
 Rajasekaran, Periagaram K., 2103 Starcrest Lane, Richardson Texas 75081,
 (US)
 McMahan, Michael L., 3817 Merriman Drive, Plano Texas 75014, (US)
 Anderson, Wallace, 1207 Edgewood Drive, Richardson Texas 75081, (US)

LEGAL REPRESENTATIVE:

Abbott, David John et al (27491), Abel & Imray Northumberland House
 303-306 High Holborn, London, WC1V 7LH, (GB)

PATENT (CC, No, Kind, Date): EP 302663 A2 890208 (Basic)
 EP 302663 A3 891011
 EP 302663 B1 931013

APPLICATION (CC, No, Date): EP 88306967 880728;

PRIORITY (CC, No, Date): US 79563 870730

DESIGNATED STATES: DE; FR; GB; IT

INTERNATIONAL PATENT CLASS: G10L-005/06

ABSTRACT WORD COUNT: 77

LANGUAGE (Publication,Procedural,Application): English; English; English
 FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS B	(English)	EPBBF1	892
CLAIMS B	(German)	EPBBF1	880
CLAIMS B	(French)	EPBBF1	1055
SPEC B	(English)	EPBBF1	3721
Total word count - document A			0
Total word count - document B			6548
Total word count - documents A + B			6548

INTERNATIONAL PATENT CLASS: G10L-005/06

...SPECIFICATION enrollment. Some variation can be tolerated in most applications. In the preferred embodiment, enrollment of words having a

length less than the expected length by up to 4 frames is considered acceptable. When a shorter word is enrolled, the silence at the end is not included in the reference template, so that the template itself is shorter than was originally expected. If the enrolled word is longer than expected...

...in the art. For example, it is possible to derive the cepstral coefficients of each frame directly, instead of performing the LPC transform first. Other transforms than the cepstrum can be used. For example, the LPC parameters could be made binary valued directly, although experimentation has indicated that the second transform into cepstral parameters yields better recognition in most instances. Also, principle spectral components can be used to generate...

14/3,K/7 (Item 1 from file: 349)
DIALOG(R) File 349:PCT FULLTEXT
(c) 2003 WIPO/Univentio. All rts. reserv.

01010960 **Image available**
IMPROVE SPEECH RECOGNITION BY DYNAMICAL NOISE MODEL ADAPTATION
RECONNAISSANCE VOCALE AMELIOREE PAR ADAPTATION DE MODELE SONORE DYNAMIQUE

Patent Applicant/Assignee:

MOTOROLA INC, 1303 East Algonquin Road, Schaumburg, IL 60196, US, US
(Residence), US (Nationality)

Inventor(s):

MA Changxue, 4929 Lichfield Drive, Barrington, IL 60010, US,
WEI Yuan-Jun, 969 Freeman Road, Hoffman Estates, IL 60195, US,

Legal Representative:

NICHOLS Daniel K (et al) (agent), Motorola, Inc., Intellectual Property
Dept., 1303 East Algonquin Road, Schaumburg, IL 60196, US,

Patent and Priority Information (Country, Number, Date):

Patent: WO 200341052 A1 20030515 (WO 0341052)

Application: WO 2002US34471 20021028 (PCT/WO US0234471)

Priority Application: US 20017886 20011105

Designated States: AE AG AL AM AT AU AZ BA BB BG BR BY BZ CA CH CN CO CR CU
CZ DE DK DM DZ EC EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP
KR KZ LC LK LR LS LT LU LV MA MD MG MK MN MW MX MZ NO NZ OM PH PL PT RO
RU SD SE SG SI SK SL TJ TM TN TR TT TZ UA UG UZ VC VN YU ZA ZM ZW
(EP) AT BE BG CH CY CZ DE DK EE ES FI FR GB GR IE IT LU MC NL PT SE SK TR
(OA) BF BJ CF CG CI CM GA GN GQ GW ML MR NE SN TD TG
(AP) GH GM KE LS MW MZ SD SL SZ TZ UG ZM ZW
(EA) AM AZ BY KG KZ MD RU TJ TM

Publication Language: English

Filing Language: English

Fulltext Word Count: 7701

Main International Patent Class: G10L-015/04

International Patent Class: G10L-015/14 ...

... G10L-015/20

Fulltext Availability:

Detailed Description

Detailed Description

... ASR art for finding the most probable sequence of HMM states.

FIG. 4 is a first part of flow chart of a process 400 for extracting feature ...applied to the log magnitude MEL scale frequency components for each frame to obtain a cepstral coefficient vector

for each frame . In step 504 first or higher order differences are taken between corresponding cepstral coefficients for two or more frames to obtain at least first order inter frame cepstral coefficient differences (deltas). In step 506 for each frame the cepstral coefficients and the inter frame cepstral coefficient differences are output as a feature vector.

FIG. 6 is a hardware block diagram of...

14/3,K/8 (Item 2 from file: 349)
DIALOG(R) File 349:PCT FULLTEXT
(c) 2003 WIPO/Univentio. All rts. reserv.

00901427 **Image available**
SPEECH RECOGNITION USING WORD-IN-PHRASE COMMAND
RECONNAISSANCE DE LA PAROLE FAISANT APPEL A UNE INSTRUCTION DE MOT DANS UNE
PHRASE

Patent Applicant/Assignee:

L & H HOLDING USA INC, 52 Third Avenue, Burlington, MA 01803, US, US
(Residence), US (Nationality), (For all designated states except: US)

Patent Applicant/Inventor:

VAN EVEN Stijn, 32 Cedarwood Road, Jamaica Plain, MA 02130, US, US
(Residence), BE (Nationality), (Designated only for: US)
LI Li, 604 Village Road East, Norwood, MA 020262, US, US (Residence), CN
(Nationality), (Designated only for: US)
DU Xianju, 74 Harvard Street #4, Waltham, MA 02453, US, US (Residence),
CN (Nationality), (Designated only for: US)
ZHAN Puming, 5 Fischer Path, Acton, MA 01720, US, US (Residence), CN
(Nationality), (Designated only for: US)

Legal Representative:

HAYDEN John F (agent), Fish & Richardson P.C., 601 Thirteenth Street, NW,
Washington, DC 20005, US,

Patent and Priority Information (Country, Number, Date):

Patent: WO 200235519 A1 20020502 (WO 0235519)
Application: WO 2001US42784 20011026 (PCT/WO US0142784)
Priority Application: US 2000696685 20001026

Parent Application/Grant:

Related by Continuation to: US 2000696685 20001026 (CON)

Designated States: AE AG AL AM AT AU AZ BA BB BG BR BY BZ CA CH CN CO CR CU
CZ DE DK DM DZ EC EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP
KR KZ LC LK LR LS LT LU LV MA MD MG MK MN MW MX MZ NO NZ OM PH PL PT RO
RU SD SE SG SI SK SL TJ TM TR TT TZ UA UG US UZ VN YU ZA ZW
(EP) AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE TR
(OA) BF BJ CF CG CI CM GA GN GQ GW ML MR NE SN TD TG
(AP) GH GM KE LS MW MZ SD SL SZ TZ UG ZW
(EA) AM AZ BY KG KZ MD RU TJ TM

Publication Language: English

Filing Language: English

Fulltext Word Count: 13508

Main International Patent Class: G10L-015/26

Fulltext Availability:

Detailed Description

Detailed Description

... the utterance.

As shown in Fig. 3, the front end processing module 200 produces a frame

from digital samples according to a procedure 300. The module first produces a frequency domain representation $X(f)$ of the portion of the utterance by performing From the normalized results, the module performs cepstral analysis to produce twelve **cepstral parameters** (step 325). The module generates the **cepstral parameters** by performing an inverse cosine transformation on the logarithms of the frequency **parameters**. **Cepstral parameters** and **cepstral differences** (described below) have been found to emphasize information important to speech recognition more effectively than

6

do the frequency parameters. After performing channel normalization of the **cepstral parameters** (step 330), the module produces twelve **cepstral differences** (that is, the differences between **cepstral parameters** in successive **frames**) (step 335) and twelve **cepstral second differences** (that is, the differences between **cepstral differences** in successive **frames**) (step 340). Finally, the module performs an IMELDA linear combination transformation to select the twenty-four most useful **parameters** from the twelve **cepstral parameters**, the twelve **cepstral differences**, and the twelve **cepstral second differences** (step 345).

Referring again to Fig. 2, a recognizer 215 receives and processes the...

14/3,K/9 (Item 3 from file: 349)

DIALOG(R)File 349:PCT FULLTEXT

(c) 2003 WIPO/Univentio. All rts. reserv.

00852962 **Image available**

VOICE ACTIVITY DETECTION AND END-POINT DETECTION
DETECTION D'ACTIVITE VOCALE ET D'EXTREMITE DE MOT

Patent Applicant/Assignee:

MULTIMEDIA TECHNOLOGIES INSTITUTE - MTI S R L, Via G. Leopardi, 41,
I-95127 Catania, IT, IT (Residence), IT (Nationality), (For all
designated states except: US)

Patent Applicant/Inventor:

BERITELLI Francesco, Multimedia Technologies Institute - MTI S.r.l., Via
G. Leopardi, 41, I-95127 Catania, IT, IT (Residence), IT (Nationality),
(Designated only for: US)

Legal Representative:

IANNONE Carlo Luigi (et al) (agent), Ing. Barzano & Zanardo Roma S.p.A.,
Via Piemonte, 26, I-00187 Roma, IT,

Patent and Priority Information (Country, Number, Date):

Patent: WO 200186633 A1 20011115 (WO 0186633)

Application: WO 2001IT221 20010508 (PCT/WO IT0100221)

Priority Application: IT 2000RM248 20000510

Designated States: AE AG AL AM AT AU AZ BA BB BG BR BY BZ CA CH CN CO CR CU
CZ DE DK DM DZ EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR
KZ LC LK LR LS LT LU LV MA MD MG MK MN MW MX MZ NO NZ PL PT RO RU SD SE
SG SI SK SL TJ TM TR TT TZ UA UG US UZ VN YU ZA ZW
(EP) AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE TR
(OA) BF BJ CF CG CI CM GA GN GW ML MR NE SN TD TG
(AP) GH GM KE LS MW MZ SD SL SZ TZ UG ZW
(EA) AM AZ BY KG KZ MD RU TJ TM

Publication Language: English

Filing Language: Italian

Fulltext Word Count: 11351

Main International Patent Class: G10L-011/02

Fulltext Availability:

Claims

English Abstract

...voice activity or VAD method in a voice signal, particularly in telephonic applications, comprising: a **first** step aimed at acquiring the voice signal (1) divided in segments or **frames** having a time durationd, a **second** step aimed at computing, for each **frame**, at least three of the following five parameters: the energy differential over the whole band...

...over the band 0-1kHz, DeltaE"sub"1, the zero crossing rate differential, DeltaZCR, the **second cepstral coefficient**, c"sub"2, and the **fifth cepstral coefficient**, c"sub"5, a third step in which a neural network process is carried out in order to provide, based upon at least three of said five parameters, for each **frame**, an output value Y in the range defined by a minimum value Y"sub"min...

Claim

... over the band 0 - 1 kHz, AE,,
the zero crossing rate differential, AZCR,
the second **cepstral coefficient**, d2, and
the fifth **cepstral coefficient**, C5,
- a third step in which a neural network process is carried out in
order to provide, based upon al least three of said five parameters, for
each **frame**, an output value Y in the range defined by a minimum value
Y,,,(inverted exclamation...

14/3,K/10 (Item 4 from file: 349)
DIALOG(R)File 349:PCT FULLTEXT
(c) 2003 WIPO/Univentio. All rts. reserv.

00799962 **Image available**

SPEECH PARAMETER COMPRESSION
COMPRESSION DE PARAMETRES RELATIFS A LA PAROLE

Patent Applicant/Assignee:

NOKIA MOBILE PHONES LIMITED, Keilalahdentie 4, FIN-02150 Espoo, FI, FI
(Residence), FI (Nationality)

Inventor(s):

TIAN Jilei, Insinoorinkatu 60 D 263, FIN-33720 Tampere, FI,
LAURILA Kari, Impivaarankatu 5, FIN-33820 Tampere, FI,

Legal Representative:

HAWS Helen (et al) (agent), Nokia IPR Department, Summit Avenue,
Farnborough, Hampshire GU14 0NG, GB,

Patent and Priority Information (Country, Number, Date):

Patent: WO 200133550 A1 20010510 (WO 0133550)

Application: WO 2000EP10299 20001019 (PCT/WO EP0010299)

Priority Application: GB 9925676 19991029

Designated States: AE AG AL AM AT AU AZ BA BB BG BR BY BZ CA CH CN CR CU CZ
DE DK DM DZ EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ
LC LK LR LS LT LU LV MA MD MG MK MN MW MX MZ NO NZ PL PT RO RU SD SE SG
SI SK SL TJ TM TR TT TZ UA UG UZ VN YU ZA ZW
(EP) AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE
(OA) BF BJ CF CG CI CM GA GN GW ML MR NE SN TD TG
(AP) GH GM KE LS MW MZ SD SL SZ TZ UG ZW
(EA) AM AZ BY KG KZ MD RU TJ TM

Publication Language: English

Filing Language: English

Fulltext Word Count: 9170

Main International Patent Class: G10L-015/02

Fulltext Availability:
Detailed Description

Detailed Description

... to the back end every time a new cepstrum is calculated i.e. every speech **frame** is processed to form a feature vector. Often, additional information concerning the time derivatives of...

...MFCC is also provided. For example, a feature vector may also contain information about the **first** and **second** time-derivatives of each **cepstral coefficient**. A conventional method for incorporating temporal information into speech vectors is to apply linear regression to a series of successive **cepstral coefficients** to generate **first** and **second** difference cepstra, referred to as 'delta' and 'delta-delta' cepstra (as indicated in the dashed...).

14/3,K/11 (Item 5 from file: 349)
DIALOG(R)File 349:PCT FULLTEXT
(c) 2003 WIPO/Univentio. All rts. reserv.

00796310 **Image available**

NATURAL LANGUAGE INTERFACE CONTROL SYSTEM
SYSTEME DE COMMANDE D'INTERFACE EN LANGAGE NATUREL

Patent Applicant/Assignee:

SONY ELECTRONICS INC, 1 Sony Drive, Park Ridge, NJ 07656, US, US
(Residence), US (Nationality)

Inventor(s):

KONOPKA Courtney Charles, 2399 Jefferson Street #20, Carlsbad, CA 92008,
US,

ALMSTRAND Lars Cristian, 1152 Oliver Avenue #8, San Diego, CA 92109, US,

Legal Representative:

FROMMER William S (agent), Frommer Lawrence & Haug LLP, 745 Fifth Avenue,
New York, NY 10151, US,

Patent and Priority Information (Country, Number, Date):

Patent: WO 200129823 A1 20010426 (WO 0129823)

Application: WO 2000US29036 20001019 (PCT/WO US0029036)

Priority Application: US 99160281 19991019

Designated States: AE AG AL AM AT AU AZ BA BB BG BR BY BZ CA CH CN CR CU CZ
DE DK DM DZ EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ
LC LK LR LS LT LU LV MA MD MG MK MN MW MX MZ NO NZ PL PT RO RU SD SE SG
SI SK SL TJ TM TR TT TZ UA UG UZ VN YU ZA ZW
(EP) AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE
(OA) BF BJ CF CG CI CM GA GN GW ML MR NE SN TD TG
(AP) GH GM KE LS MW MZ SD SL SZ TZ UG ZW
(EA) AM AZ BY KG KZ MD RU TJ TM

Publication Language: English

Filing Language: English

Fulltext Word Count: 11226

Main International Patent Class: G10L-015/22

Fulltext Availability:

Detailed Description

Detailed Description

... attention words (Step 402). This allows the NLICS to accept

non-prompted user requests, but **first** the system must be told that a user request is coming. The attention word accomplishes...

...dimensional feature vector is derived from the acoustic data. These features consist of Mel-Frequency Cepstral coefficients 1-12 and the **first** and **second** order derivatives of MFC coefficients 0. Thus, feature vectors are created from the acoustic data...

14/3, K/12 (Item 6 from file: 349)

DIALOG(R) File 349:PCT FULLTEXT
(c) 2003 WIPO/Univentio. All rts. reserv.

00566699 **Image available**

MITIGATING ERRORS IN A DISTRIBUTED SPEECH RECOGNITION PROCESS
ATTENUATION DES ERREURS DANS UN PROCESSUS DE RECONNAISSANCE DE LA PAROLE
REPARTI

Patent Applicant/Assignee:

MOTOROLA LIMITED,
PEARCE David John Benjamin,
GIBBS Jon Alastair,

Inventor(s):

PEARCE David John Benjamin,
GIBBS Jon Alastair,

Patent and Priority Information (Country, Number, Date):

Patent: WO 200030072 A1 20000525 (WO 0030072)
Application: WO 99EP9028 19991112 (PCT/WO EP9909028)
Priority Application: GB 9824894 19981113

Designated States: AE AL AM AT AU AZ BA BB BG BR BY CA CH CN CU CZ DE DK EE
ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR LS LT
LU LV MD MG MK MN MW NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT
UA UG US UZ VN YU ZA ZW AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL
PT SE

Publication Language: English

Fulltext Word Count: 7218

Main International Patent Class: G10L-015/26

Fulltext Availability:

Detailed Description

Detailed Description

... of speech recognition
parameters other than cepstral coefficients.

The fourteen parameters for each sampling time- **frame** are arranged, or formatted, into a corresponding vector, also known as an array, as shown in

FIG. 1. Vector 131 corresponds to sampling time- **frame** 121, vector 132 corresponds to sampling time- **frame** 122, vector 133 corresponds to sampling time- **frame** 123, and vector 134 corresponds to sampling time- **frame** 124. Such a vector can generally be represented as

$F c(m) I$
 $Y(M\dots)$

...10g[E(m)@

5

The speech recognition parameters are processed prior to transmission from a **first** location to a **second** location. In the embodiment described below this is carried out as follows. The parameters from...

14/3,K/13 (Item 7 from file: 349)
DIALOG(R)File 349:PCT FULLTEXT
(c) 2003 WIPO/Univentio. All rts. reserv.

00551351 **Image available**
METHOD FOR REDUCING NOISE DISTORTIONS IN A SPEECH RECOGNITION SYSTEM
PROCEDE POUR REDUIRE DES DISTORSIONS DE BRUIT DANS UN SYSTEME DE
RECONNAISSANCE VOCALE

Patent Applicant/Assignee:
SONY ELECTRONICS INC,

Inventor(s):

MENENDEZ-PIDAL Xavier,
TANAKA Miyuki,
CHEN Ruxin,
WU Duanpei,

Patent and Priority Information (Country, Number, Date):

Patent: WO 200014724 A1 20000316 (WO 0014724)

Application: WO 99US18896 19990819 (PCT/WO US9918896)

Priority Application: US 9899537 19980909; US 98177461 19981022

Designated States: AE AL AM AT AU AZ BA BB BG BR BY CA CH CN CU CZ DE DK EE
ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR LS LT
LU LV MD MG MK MN MW NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT
UA UG UZ VN YU ZA ZW GH GM KE LS MW SD SL SZ UG ZW AM AZ BY KG KZ MD RU
TJ TM AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE BF BJ CF CG
CI CM GA GN GW ML MR NE SN TD TG

Publication Language: English

Fulltext Word Count: 8737

Main International Patent Class: G10L-003/02

International Patent Class: G10L-009/00

Fulltext Availability:

Claims

Claim

... said channels in said logarithmic channel energy.

14 The system of claim 12 wherein a first time cosine transform (536) converts said static features (534) into delta features (542), and wherein...

...features (534) into delta-delta features (544).

22

. The system of claim 14 wherein said first time cosine transform (536) and said second time cosine transform (536) each perform a centered...

...P)Cos@ O;T

at k=-M 2M+I

where Ct(p) is a pth cepstral coefficient at a time frame t, M is half of a window size used to estimate differential coefficients, and o ... I C,+k (P

at k=-M 2M+I

where Ct(p) is a pth cepstral coefficient at a time frame t, M is half of a window size used to estimate differential coefficients, and o

...

14/3,K/14 (Item 8 from file: 349)

DIALOG(R)File 349:PCT FULLTEXT

(c) 2003 WIPO/Univentio. All rts. reserv.

00540013 **Image available**

LANGUAGE INDEPENDENT SPEECH RECOGNITION
RECONNAISSANCE DE LA PAROLE INDEPENDANTE DE LA LANGUE

Patent Applicant/Assignee:

LERNOUT & HAUSPIE SPEECH PRODUCTS N V,

Inventor(s):

D'HOORE Bart,

VAN COMPERNOLLE Dirk,

Patent and Priority Information (Country, Number, Date):

Patent: WO 200003386 A1 20000120 (WO 0003386)

Application: WO 99IB1406 19990708 (PCT/WO IB9901406)

Priority Application: US 98113589 19980710.

Designated States: JP AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE

Publication Language: English

Fulltext Word Count: 4623

Main International Patent Class: G10L-015/00

Fulltext Availability:

Detailed Description

Detailed Description

... a telephone speech recognition engine uses the commonly known LPC analysis method to derive 12 **cepstral coefficients** and log 1 5 energy, along with **first** and **second** order derivatives. A preferred embodiment of a microphone speech recognition engine uses the commonly known...

...FFT method to accomplish the same purpose. The result for both engines for each speech **frame** is a vector of 12 cepstra, 12 delta cepstra, 12 delta delta cepstra, delta log...

14/3,K/15 (Item 9 from file: 349)

DIALOG(R)File 349:PCT FULLTEXT

(c) 2003 WIPO/Univentio. All rts. reserv.

00237266

PRIORITIZATION METHOD AND DEVICE FOR SPEECH FRAMES CODED BY A LINEAR PREDICTIVE CODER

PROCEDE ET DISPOSITIF D'ATTRIBUTION DE PRIORITE POUR BLOCS DE SIGNAUX VOCAUX A L'AIDE D'UN CODEUR A PREDICTION LINEAIRE

Patent Applicant/Assignee:

MOTOROLA INC,

Inventor(s):

YONG Mei,

Patent and Priority Information (Country, Number, Date):

Patent: WO 9311530 A1 19930610

Application: WO 92US8053 19920921 (PCT/WO US9208053)

Priority Application: US 91881 19911126

Designated States: AU CA JP AT BE CH DE DK ES FR GB GR IE IT LU MC NL SE

Publication Language: English

Fulltext Word Count: 9393

Main International Patent Class: G10L-009/00

Fulltext Availability:

Claims

Claim

... the steps of:
4A) initializing to desired settings a memory unit having at least a **first** memory location (M1) for onset condition storage of an immediately preceding speech frame (IPSF) and...CSF and for determining an onset condition of the CSF;
4E) utilizing the at least **first** and second memory locations for storing the onset condition of the CSF, the LPC coefficients...of the CSF indicates an onset speech frame, setting the IPSF onset condition in the **first** memory location to ONSET; and
412) where the onset condition of the CSF indicates a non-onset speech frame, setting the IPSF onset condition in the **first** memory location to NON-ONSET,
4J) and, where selected, wherein at least one of 4J1...

...ONSET;
4J2) the log spectral distance is determined by determining a mean squared error of **cepstral coefficients** I 0 between the selected current **frame** and its immediately preceding **frame** ,, the **cepstral coefficients** for a speech **frame** being determined iteratively from the LPC coefficients and prediction error energy for the CSF;
4J3...

14/3,K/16 (Item 10 from file: 349)
DIALOG(R) File 349:PCT FULLTEXT
(c) 2003 WIPO/Univentio. All rts. reserv.

00209265
METHODS AND APPARATUS FOR VERIFYING THE ORIGINATOR OF A SEQUENCE OF OPERATIONS
METHODES ET APPAREILS POUR VERIFIER L'IDENTITE DE L'INITIATEUR D'UNE SEQUENCE D'OPERATIONS

Patent Applicant/Assignee:

ENSIGMA LIMITED,
THE SECRETARY OF STATE FOR DEFENCE,
CAREY Michael John,
PARRIS Eluned Sarah,
BRIDLE John Scott,

Inventor(s):

CAREY Michael John,
PARRIS Eluned Sarah,
BRIDLE John Scott,

Patent and Priority Information (Country, Number, Date):

Patent: WO 9206468 A1 19920416
Application: WO 91GB1681 19910930 (PCT/WO GB9101681)
Priority Application: GB 9021489 19901003

Designated States: AT AU BE CH DE DK ES FR GB GR IT JP LU NL SE US

Publication Language: English

Fulltext Word Count: 8219

Main International Patent Class: G10L-005/06

Fulltext Availability:

Detailed Description

Detailed Description

... effect of telephone line distortion on the spectrum of the speech signal, the zero and **first** **cepstral coefficients** , known

as MFCC0 and MFCC1 may be given zero weights. Processing preferably also includes deriving an indication related to the rate of change of each coefficient (**first** order difference) and its **second** order difference. An algorithm for this purpose (see Figure 4) begins by setting the number ...an operation 52 and setting (operation 53) a variable **k** nominally representing a particular recent **frame** to -kmax where kmax is the number of previous and succeeding **frames** relative to the **frame** **k** to be used in forming each jth **first** order rate of difference dj.

An operation 54 and a test 55 cause (2kmax + 1...

?

19/3,K/1 (Item 1 from file: 348)
DIALOG(R)File 348:EUROPEAN PATENTS
(c) 2003 European Patent Office. All rts. reserv.

00208476

Method of and system for speech recognition.

Verfahren und Einrichtung zur Spracherkennung.

Procede et dispositif de reconnaissance de la parole.

PATENT ASSIGNEE:

Oki Electric Industry Company, Limited, (225690), 7-12, Toranomon 1-chome
Minato-ku, Tokyo 105, (JP), (applicant designated states: DE;FR;GB;NL)

INVENTOR:

Morito, Makoto c/o Oki Electr. Ind. Co., Ltd., 7-12, Toranomon 1-chome,
Minato-ku Tokyo, (JP)

Tabei, Yukio c/o Oki Electr. Ind. Co., Ltd., 7-12, Toranomon 1-chome,
Minato-ku Tokyo, (JP)

Yamada, Kozo c/o Oki Electr. Ind. Co., Ltd., 7-12, Toranomon 1-chome,
Minato-ku Tokyo, (JP)

LEGAL REPRESENTATIVE:

Betten, Jürgen, Dipl.-Ing. et al (38515), Patentanwalte Betten & Resch
Reichenbachstrasse 19, W-8000 München 5, (DE)

PATENT (CC, No, Kind, Date): EP 219712 A1 870429 (Basic)
EP 219712 B1 920108

APPLICATION (CC, No, Date): EP 86113175 860925;

PRIORITY (CC, No, Date): JP 85213417 850926; JP 85213418 850926; JP
85224878 851011; JP 86451 860106

DESIGNATED STATES: DE; FR; GB; NL

INTERNATIONAL PATENT CLASS: G10L-007/08;

ABSTRACT WORD COUNT: 241

LANGUAGE (Publication, Procedural, Application): English; English; English
FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS B	(English)	EPBBF1	2929
CLAIMS B	(German)	EPBBF1	1859
CLAIMS B	(French)	EPBBF1	2630
SPEC B	(English)	EPBBF1	7955
Total word count - document A			0
Total word count - document B			15373
Total word count - documents A + B			15373

...SPECIFICATION vectors as created by the filter banks.

The same is also valid for the local **peak** detection method known from
ICASSP 84 proceedings, volume 1, pages 9.9.1 to 9.9.4, which is also
not applicable to real- time speech recognition systems.

In view of the drawbacks of the prior speech recognition techniques, it
...means are respectively realizable with exclusive hardware without use
of the processor.

The above and **other** objects, features and advantages of the present
invention will become more apparent from the following description when
taken in **conjunction** with the accompanying drawings in which a
preferred embodiment of the present invention is shown by way of
illustrative example.

Fig. 1 is a flowchart illustrating processing by a prior speech
recognition method;

Fig. 2 is a block diagram illustrating a speech recognition
apparatus according to the present invention;

Fig. 3 is a view illustrating a characteristic of a band pass
filter of a signal processor 15...

19/3,K/2 (Item 1 from file: 349)

DIALOG(R)File 349:PCT FULLTEXT

(c) 2003 WIPO/Univentio. All rts. reserv.

00806382

METHOD FOR AFFORDING A MARKET SPACE INTERFACE BETWEEN A PLURALITY OF MANUFACTURERS AND SERVICE PROVIDERS AND INSTALLATION MANAGEMENT VIA A MARKET SPACE INTERFACE

PROCEDE DE MISE A DISPOSITION D'UNE INTERFACE D'ESPACE DE MARCHE ENTRE UNE PLURALITE DE FABRICANTS ET DES FOURNISSEURS DE SERVICES ET GESTION D'UNE INSTALLATION VIA UNE INTERFACE D'ESPACE DE MARCHE

Patent Applicant/Assignee:

ACCENTURE LLP, 1661 Page Mill Road, Palo Alto, CA 94304, US, US
(Residence), US (Nationality)

Inventor(s):

MIKURAK Michael G, 108 Englewood Blvd., Hamilton, NJ 08610, US,

Legal Representative:

HICKMAN Paul L (et al) (agent), Oppenheimer Wolff & Donnelly LLP, 1400 Page Mill Road, Palo Alto, CA 94304, US,

Patent and Priority Information (Country, Number, Date):

Patent: WO 200139028 A2 20010531 (WO 0139028)

Application: WO 2000US32308 20001122 (PCT/WO US0032308)

Priority Application: US 99444773 19991122; US 99444798 19991122

Designated States: AE AG AL AM AT AU AZ BA BB BG BR BY BZ CA CH CN CR CU CZ
DE DK DM DZ EE ES FI GB GE GH GM HR HU ID IL IS JP KE KG KP KR KZ LC LK
LR LS LT LU LV MA MD MG MK MN MW MX MZ NO NZ PL PT RO RU SD SE SG SI SK
SL TJ TM TR TT TZ UA UG UZ VN YU ZW

(EP) AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE TR

(OA) BF BJ CF CG CI CM GA GN GW ML MR NE SN TD TG

(AP) GH GM KE LS MW MZ SD SL SZ TZ UG ZW

(EA) AM AZ BY KG KZ MD RU TJ TM

Publication Language: English

Filing Language: English

Fulltext Word Count: 170977

Fulltext Availability:

Detailed Description

Detailed Description

... the fact that not all electric power costs the same to generate. Power generated during **peak** times is more expensive than "base-line" power. For demand side management, utility companies will...themselves typically are referred to as users, in the context of the network. Blocks or **frames** of data are transmitted over a link along a path between nodes of the network...network. The three layers of the X.25 interface architecture are the physical level, the **frame** level and the packet level. Although data communication between DCEs of the network is routinely...years. NINS became the focus of service providers in 1995 as they saw revenues for **frame** relay network services double for two years in a row. What began as a way to boost the popularity of **frame** relay services by offering to lease and manage routers has blossomed into a diverse set...

...end of the continuum consists of NINS for cur-rent network services, including leased lines, **frame** relay, and X On the far end is outsourced MNS characterized by long-term contracts...Core" Network Architecture The current wire-line "Core" network consists of parallel PSTN, SMDS, ATM, **Frame** -Relay, B/PRI and LP networks. The PSTN network has been evolving over the last...3/STM-1).

ryi

The data networks consist of many technologies e.g. SMDS, ATM, **frame** -relay and IP.

In some cases, these data networks themselves are parallel networks, in other...

...share a common technology in the backbone (e.g. ATM can be the backbone for **frame** relay and IP data networks). These data networks share the same SONET based backbone with...

?

24/3,K/1 (Item 1 from file: 348)

DIALOG(R) File 348:EUROPEAN PATENTS

(c) 2003 European Patent Office. All rts. reserv.

00208476

Method of and system for speech recognition.

Verfahren und Einrichtung zur Spracherkennung.

Procede et dispositif de reconnaissance de la parole.

PATENT ASSIGNEE:

Oki Electric Industry Company, Limited, (225690), 7-12, Toranomon 1-chome
Minato-ku, Tokyo 105, (JP), (applicant designated states: DE;FR;GB;NL)

INVENTOR:

Morito, Makoto c/o Oki Electr. Ind. Co., Ltd., 7-12, Toranomon 1-chome,
Minato-ku Tokyo, (JP)

Tabei, Yukio c/o Oki Electr. Ind. Co., Ltd., 7-12, Toranomon 1-chome,
Minato-ku Tokyo, (JP)

Yamada, Kozo c/o Oki Electr. Ind. Co., Ltd., 7-12, Toranomon 1-chome,
Minato-ku Tokyo, (JP)

LEGAL REPRESENTATIVE:

Betten, Jürgen, Dipl.-Ing. et al (38515), Patentanwälte Betten & Resch
Reichenbachstrasse 19, W-8000 München 5, (DE)

PATENT (CC, No, Kind, Date): EP 219712 A1 870429 (Basic)
EP 219712 B1 920108

APPLICATION (CC, No, Date): EP 86113175 860925;

PRIORITY (CC, No, Date): JP 85213417 850926; JP 85213418 850926; JP
85224878 851011; JP 86451 860106

DESIGNATED STATES: DE; FR; GB; NL

INTERNATIONAL PATENT CLASS: G10L-007/08;

ABSTRACT WORD COUNT: 241

LANGUAGE (Publication,Procedural,Application): English; English; English
FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS B	(English)	EPBBF1	2929
CLAIMS B	(German)	EPBBF1	1859
CLAIMS B	(French)	EPBBF1	2630
SPEC B	(English)	EPBBF1	7955
Total word count - document A			0
Total word count - document B			15373
Total word count - documents A + B			15373

...SPECIFICATION a third embodiment of the present invention;

Fig. 5 (A) is a view exemplarily illustrating **frame** power P_i
under noiseless environment;

Fig. 5 (B) is a view exemplarily illustrating **frame** power P_i'
under noisy environment due to automobiles;

Fig. 6 (A) to (C) are respectively views illustrating **evaluation**
of a local peaks vector;

Fig. 7 is a view illustrating evaluation for pattern similarity...

...Fig. 9 is a view illustrating linear expansion of a time axis of an
input **pattern**;

Fig. 10 is a flowchart illustrating **evaluation** for a local **peaks**
vector in the third and fourth embodiments of the present invention;
and

Figs. 11(A) to (E) are respectively views illustrating the
evaluation for...
m designates the number of a reference pattern while j
the number of a speech **frame**. For simplifying the description, a **start**
point of the learning speech **yielded** in the voiced interval detection
(S13) is assumed to be 1 while an end point...

...preparing the reference pattern as described above was designated as the registration processing, processing of **recognizing** an input **speech** is designated as **recognition** processing. Thereupon, any **speech** input in the **recognition** processing is called an input speech. For this input speech, the numbers of speech frames...the first through fourth embodiments described previously was performed by the nonlinear matching using the **dynamic programming** method, but this evaluation can be performed by means of the linear matching. When performing...
...input speech is subjected to time axis linear expansion or compression into a prescribed speech **frames** length.

"Linear expansion or compression"

The linear expansion or **compression** processing is mainly to facilitate linear matching described later and is furthermore to facilitate area control upon...

...time axis linear expansion will be described. A case is described for brevity where input **local peaks** vectors are linearly expanded into thirty two speech **frames**. The start point and the end point are respectively assumed to be S and E...

...speech frame number after the linear expansion is assumed to be i' ($i' = 1$ through 32), and a speech **frame** number before the linear expansion or compression is assumed to be i . The speech frame...

...19): (see image in original document)
and an input local peaks vector R_i in the i th speech frame before the linear expansion or compression is assumed to be an input local peaks ...speech feature vectors shown in the second and fourth embodiments described previously, whereby highly accurate **speech recognition** can be assured. Furthermore, as described in the third embodiment, according to the present invention, local **peaks** vectors are evaluated by estimating window vectors from **feature** vectors after **spectrum** normalization of an input speech, smoothing the window vectors, and multiplying the feature vector after...

...spectrum window. Accordingly no erroneous discrimination is produced between a local peak due to noises and that due to a speech for assuring highly accurate processings in the similarity evaluation with each reference pattern...

...judgement thereof.

As clearly evidenced from the above description, according to the present invention, a **speech recognition** system with excellent **recognition** accuracy can be assured.

From the foregoing, it will now be apparent that new and improved **speech recognition** method and system have been found. It should be understood of course that the embodiments...

...CLAIMS name given to a reference pattern having a maximum pattern similarity among said pattern similarities evaluated for each reference pattern as a **recognized** result (S19);

characterized by said step (c) for converting to said local peaks vector R_i comprising the steps of :

(A) evaluating a speech feature vector B_i by subtracting said noise pattern N from each feature vector...

...evaluating a spectrum normalized speech feature vector Z_i composed of

components z_i ($sub(i)$) ($sup(k)$), where i is a subscript indicative of a speech **frame** number and k is a superscript indicative of the channel **number** $k = 1$ to K , by spectrum normalizing said speech feature vector B_i using said least...

24/3,K/2 (Item 1 from file: 349)
DIALOG(R)File 349:PCT FULLTEXT
(c) 2003 WIPO/Univentio. All rts. reserv.

00799959 **Image available**
METHODS AND APPARATUSES FOR SIGNAL ANALYSIS
PROCEDES ET APPAREILS D'ANALYSE DE SIGNAUX
Patent Applicant/Assignee:
HUQ SPEECH TECHNOLOGIES B V, Zilverlaan 2, NL-9743 RK Groningen, NL, NL
(Residence), NL (Nationality), (For all designated states except: US)
Patent Applicant/Inventor:
ANDRINGA Tjeerd Catharinus, Paterswoldseweg 324, NL-9727 BX Groningen, NL
, NL (Residence), NL (Nationality), (Designated only for: US)
DUIFHUIS Hendrikus, Zonnebloemweg 21, NL-9765 HW Paterswolde, NL, NL
(Residence), NL (Nationality), (Designated only for: US)
VAN HENGEL Pieter Willem Jan, Hiddemaheerd 38, NL-9737 JP Groningen, NL
, NL (Residence), NL (Nationality), (Designated only for: US)
HEEMSKERK Michael Gerardus, Aquamarijnstraat 421, NL-9743 PK Groningen,
NL, NL (Residence), NL (Nationality), (Designated only for: US)
NILLESEN Maartje Marjolein, Friesestraatweg 5A, NL-9718 NA Groningen, NL,
NL (Residence), NL (Nationality), (Designated only for: US)

Legal Representative:
PRINS Ir A W (agent), Vereenigde, Nieuwe Parklaan 97, NL-2587 BN The
Hague, NL,
Patent and Priority Information (Country, Number, Date):
Patent: WO 200133547 A1 20010510 (WO 0133547)
Application: WO 2000NL808 20001106 (PCT/WO NL0000808)
Priority Application: NL 1013500 19991105
Designated States: AE AG AL AM AT AU AZ BA BB BG BR BY BZ CA CH CN CR CU CZ
DE DK DM DZ EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ
LC LK LR LS LT LU LV MA MD MG MK MN MW MX MZ NO NZ PL PT RO RU SD SE SG
SI SK SL TJ TM TR TT TZ UA UG US UZ VN YU ZA ZW
(EP) AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE TR
(OA) BF BJ CF CG CI CM GA GN GW ML MR NE SN TD TG
(AP) GH GM KE LS MW MZ SD SL SZ TZ UG ZW
(EA) AM AZ BY KG KZ MD RU TJ TM

Publication Language: English
Filing Language: English
Fulltext Word Count: 24777

Fulltext Availability:
Claims

Claim
... device can be used estimate fundamental period contour for speech databases prior to a automatic **speech recognition**. The demands for a fundamental period estimation algorithm to measure the robustness of a **speech recognition** system are slightly different from a system that aims to select and track as much...

...therefore an approximation. The optimization in the selection algorithm (see Selection of periodic signal contributions) **determines** the final instantaneous fundamental period.

In each **frame** the three highest **peaks** in the summed autocorrelation

with values higher than 0.3 times the local energy along...

?

File 9:Business & Industry(R) Jul/1994-2003/Dec 02
(c) 2003 Resp. DB Svcs.
File 15:ABI/Inform(R) 1971-2003/Dec 02
(c) 2003 ProQuest Info&Learning
File 16:Gale Group PROMT(R) 1990-2003/Dec 02
(c) 2003 The Gale Group
File 20:Dialog Global Reporter 1997-2003/Dec 03
(c) 2003 The Dialog Corp.
File 47:Gale Group Magazine DB(TM) 1959-2003/Dec 02
(c) 2003 The Gale group
File 75:TGG Management Contents(R) 86-2003/Nov W3
(c) 2003 The Gale Group
File 80:TGG Aerospace/Def.Mkts(R) 1986-2003/Dec 02
(c) 2003 The Gale Group
File 88:Gale Group Business A.R.T.S. 1976-2003/Dec 02
(c) 2003 The Gale Group
File 98:General Sci Abs/Full-Text 1984-2003/Oct
(c) 2003 The HW Wilson Co.
File 112:UBM Industry News 1998-2003/Dec 03
(c) 2003 United Business Media
File 141:Readers Guide 1983-2003/Oct
(c) 2003 The HW Wilson Co
File 148:Gale Group Trade & Industry DB 1976-2003/Dec 02
(c) 2003 The Gale Group
File 160:Gale Group PROMT(R) 1972-1989
(c) 1999 The Gale Group
File 275:Gale Group Computer DB(TM) 1983-2003/Dec 02
(c) 2003 The Gale Group
File 264:DIALOG Defense Newsletters 1989-2003/Dec 02
(c) 2003 The Dialog Corp.
File 484:Periodical Abs Plustext 1986-2003/Nov W4
(c) 2003 ProQuest
File 553:Wilson Bus. Abs. FullText 1982-2003/Oct
(c) 2003 The HW Wilson Co
File 570:Gale Group MARS(R) 1984-2003/Dec 02
(c) 2003 The Gale Group
File 608:KR/T Bus.News. 1992-2003/Dec 03
(c) 2003 Knight Ridder/Tribune Bus News
File 620:EIU:Viewswire 2003/Dec 02
(c) 2003 Economist Intelligence Unit
File 613:PR Newswire 1999-2003/Dec 02
(c) 2003 PR Newswire Association Inc
File 621:Gale Group New Prod.Annou.(R) 1985-2003/Dec 02
(c) 2003 The Gale Group
File 623:Business Week 1985-2003/Dec 02
(c) 2003 The McGraw-Hill Companies Inc
File 624:McGraw-Hill Publications 1985-2003/Dec 02
(c) 2003 McGraw-Hill Co. Inc
File 634:San Jose Mercury Jun 1985-2003/Dec 02
(c) 2003 San Jose Mercury News
File 635:Business Dateline(R) 1985-2003/Dec 02
(c) 2003 ProQuest Info&Learning
File 636:Gale Group Newsletter DB(TM) 1987-2003/Dec 02
(c) 2003 The Gale Group
File 647:cmp Computer Fulltext 1988-2003/Nov W5
(c) 2003 CMP Media, LLC
File 696:DIALOG Telecom. Newsletters 1995-2003/Dec 02
(c) 2003 The Dialog Corp.
File 674:Computer News Fulltext 1989-2003/Nov W4
(c) 2003 IDG Communications
File 810:Business Wire 1986-1999/Feb 28

(c) 1999 Business Wire
File 813:PR Newswire 1987-1999/Apr 30
(c) 1999 PR Newswire Association Inc
? ds

Set	Items	Description
S1	28	CEPSTRAL(3N) (COEFFICIENT? OR PARAMETER? OR VALUES)
S2	0	(COMPAR? OR EVALUAT? OR DETERMIN? OR ASSESS? OR DISCERN? OR DISTINGUISH?) (5N)S1
S3	1230806	PEAK?
S4	20712	(MANY OR MULTIPLE OR SEVERAL OR MULTI OR NUMEROUS OR PLURAL?) (5N)S3
S5	1193201	FRAME??
S6	80474	SPEECH(3N)RECOG?
S7	169	ACOUSTIC(3N)DATA(10N) (PARTITION? OR DIVID? OR SEPARAT? OR - DIVISION? OR PART OR PARTS OR SECTION?? OR SEGMENT?? OR PORTION?? OR FRAGMENT? OR PIECES OR SECTOR??)
S8	94	AU=(SHU, C? OR SHU, H? OR SHU C? OR SHU H?)
S9	0	S6 AND S8
S10	0	S1(S)S4
S11	4	S1(S)S5
S12	3	RD S11 (unique items)
S13	0	S1 AND S8
S14	3	S5(S)S7
S15	3	S14 NOT S11
S16	1	RD S15 (unique items)
S17	95	S4(S)S5
S18	0	S17(S)(S6 OR S7)
S19	0	S17(S)CEPSTRAL

12/3,K/1 (Item 1 from file: 16)
DIALOG(R)File 16:Gale Group PROMT(R)
(c) 2003 The Gale Group. All rts. reserv.

06194576 Supplier Number: 54112823 (USE FORMAT 7 FOR FULLTEXT)
Voice security compares features. (Layered Biometric Verification, or LBV, insures accurate verification of user identity) (Technology Information)
Declerq, Francis
Electronic Engineering Times, p58(1)
March 15, 1999
Language: English Record Type: Fulltext
Document Type: Magazine/Journal; Trade
Word Count: 1127

... extracted and compared with previously trained models. Those features are computed from regular intervals, or **frames**, of speech using a technique called "windowing." Each **frame** is reduced to around two dozen parameters. Examples of speech features include linear prediction **coefficients**, melwarped **cepstral coefficients** and fundamental frequency.

If we record speech at 11 kHz and use a 10-ms...

12/3,K/2 (Item 1 from file: 88)
DIALOG(R)File 88:Gale Group Business A.R.T.S.
(c) 2003 The Gale Group. All rts. reserv.

03059250 SUPPLIER NUMBER: 14134739
Speaker identification based on a matrix quantization method.
Chen, Ming-Shih; Lin, Pei-Hwa; Wang, Hsiao-Chuan
IEEE Transactions on Signal Processing, v41, n1, p398(6)
Jan, 1993
ISSN: 1053-587X LANGUAGE: English RECORD TYPE: Abstract

...ABSTRACT: approach is introduced where a matrix is formed from feature vectors obtained from the weighted **cepstral coefficients** of a block of **frames**. Results demonstrate attainable identification rates of 100% even for 2-s utterances.

12/3,K/3 (Item 1 from file: 647)
DIALOG(R)File 647: CMP Computer Fulltext
(c) 2003 CMP Media, LLC. All rts. reserv.

01187008 CMP ACCESSION NUMBER: EET19990315S0048
Voice security compares features
Francis Declerq, President & Chief Executive Officer, Keyware Technologies, Woburn, Mass.
ELECTRONIC ENGINEERING TIMES, 1999, n 1052, PG58
PUBLICATION DATE: 990315
JOURNAL CODE: EET LANGUAGE: English
RECORD TYPE: Fulltext
SECTION HEADING: Signals - Focus: Voice/Speech Recognition
WORD COUNT: 1136

... extracted and compared with previously trained models. Those features are computed from regular intervals, or **frames**, of speech using a technique called "windowing." Each **frame** is reduced to around two dozen parameters. Examples of speech features include linear prediction **coefficients**, melwarped **cepstral coefficients** and fundamental

frequency.

If we record speech at 11 kHz and use a 10-ms...
?

16/3,K/1 (Item 1 from file: 15)
DIALOG(R)File 15:ABI/Inform(R)
(c) 2003 ProQuest Info&Learning. All rts. reserv.

01981025 48771415
Transcribing broadcast news for audio and video indexing
Gauvain, Jean-Luc; Lamel, Lori; Adda, Gilles
Association for Computing Machinery. Communications of the ACM v43n2 PP:
64-70 Feb 2000
ISSN: 0001-0782 JRNL CODE: GACM
WORD COUNT: 3797

...TEXT: this approach the average centisecond frame level segmentation error measured on 2 hours of test data is 3.7%. A cluster of segments usually represents a speaker in a given acoustic environment. Thus, there are typically slightly more clusters than true speakers in a show.
For...
?